AETA

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SCOOP 5

Stereo audio codec for real time audio transmission

User Manual

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1. Presentation – Getting started

The SCOOP 5 codec allows the bi-directional transmission of one or two audio signals with bit rate reduction, over various transmission media: digital leased lines, ISDN lines, PSTN telephone lines, IP protocol networks…

The standard version of the codec includes an Ethernet interface for IP transmission, and X24/X21 interfaces for transmission over digital leased lines. The unit can be complemented with many options providing additional network interfaces, coding algorithms, etc.

One outstanding feature of AETA codecs in ISDN mode is the 5A System®: on receiving an incoming ISDN call, the unit can automatically detect the coding algorithm and parameters of the calling codec, and then adjust itself in a compatible configuration so that the connection succeeds regardless of the initial configuration and that of the remote unit.

In IP mode, the codec features the same ease of operation thanks to the use of the SIP and SDP protocols.

For ISDN transmission, the unit can be operated in a "double codec" mode. It is then equivalent to two independent mono codecs, each running G711 or G722 over a B channel of the ISDN (interface #1).

This chapter gives **basic instructions** for a quick start. It obviously does not provide all the information for full control. For comprehensive information one can refer to the rest of this manual:

- Chapter 2 describes all the **functions** and features of the SCOOP 5 (but not necessarily with all the operating modes)
- Chapter 3 gives a **physical description** of the unit, shows its **setting up** and **operation principles**.
- Chapter 4 details **menus** and **operating modes**.
- Chapter 5 deals with using the **html server** embedded in SCOOP 5
- Chapter 6 provides all the **technical characteristics** of the SCOOP 5
- The annexes bring miscellaneous additional information, including an **index** you can use to look for a given information topic.

The following table shows the main features of the product. Functions marked with ● in this table are available as options. Those marked with ∇ are only available in units equipped with ISDN interface(s).

® 5AS = Aeta Audio Advanced Automatic Adjustment System

Characteristics		Optional
Operation modes		
Single wide band codec		
Double 7 kHz codec (ISDN mode)		∇
IP transmission interface		
Ethernet Interface, 10BaseT / 100BaseT; TCP/IP, UDP/IP, RTP protocols		
Audio transmission in unicast mode: SIP signalling protocol, SDP, RTP streaming		
Audio transmission in multicast mode: RTP streaming		
Net bit rate 16 to 256 kbit/s (depending on coding algorithm)		
Leased line transmission interfaces		
Two X24/X21/V11/V35 interfaces;		
64, 128, 192, 256 or 384 kbit/s over one interface (selectable)		
ISDN transmission interface		●
One or two S0 interfaces		
Single codec 64, 128, 192 or 256 kbit/s, or double codec 64 kbit/s		
5AS auto configuration on incoming calls		
Transmission interface on PSTN telephone line		●
One "2 wire" telephone interface		
« POTS codec » mode with integrated V34 modem and CELP coding, 12 to 24 kbit/s		
Mobile network access		●
Integrated 2G/3G/3G+/LTE (depending on version) network access module, 2 antenna sockets		
Voice mode: standard telephone or "HD Voice" (7 kHz with AMR-WB)		
Packet data mode: IP protocol, SIP signalling, SDP, RTP streaming, net bit rate 16 to 256 kbit/s (depending on coding algorithm)		
External 3G/LTE module connection via USB socket (data mode only)		
SMS reception		
Audio coding algorithms	(audio modes)	
G711 (standard telephone)	Mono	
GSM, AMR (mobile telephone)	Mono	
AMR-WB / G722.2 (mobile "HD Voice"	Mono	
G722 SRT, H221, H242	Mono	
CELP 7 kHz	Mono	
MPEG Audio Layer II	Mono, Stereo, Double mono, Joint stereo	
MPEG AAC-LC , HE-AAC, HE-AAC v2	Mono, Stereo	●
4 sub-band ADPCM (low latency)	Mono, Stereo	
TDAC (ISDN mode only)	Mono	● ∇
Audio interfaces		
Two analog inputs and two analog outputs with adjustable gain		
Digital input and output, AES/EBU format		
Level display for encoder inputs and ecoder outputs		
Stereo headphone socket for monitoring, switchable send/receive		
Auxiliary functions (available depending on transmission interface)		
Relay transmission: 2 isolated inputs and 2 isolated outputs, 6 non isolated inputs and outputs		
Data channel with RS232 serial port, 300 to 9600 bauds		
Audio coordination channel (bandwidth 300-3400 Hz)		●
Control and supervision		
Keyboard and LCD display on front panel		
Programmable set-up/dial memories		
Ethernet/IP remote control		
Embedded html server		
Remote control serial port, isolated control and status loops		
Secondary Ethernet interface for remote control		●

Table 1 – Main features of the SCOOP 5

1.1. Install and connect SCOOP 5

- Plug on a power source: the SCOOP 5 automatically starts up. To put it on stand-by or restart it, press the ⏻ key (top left corner on the front panel) for at least 3 seconds.
- Connect the necessary audio interfaces *(details: page 29)*
- Connect SCOOP 5 on the transmission network *(details: page 29 and following)*
- Using the SCOOP 5 menus: Esc key or OK key to activate the main menu, use the arrows to select a sub-menu or a parameter, enter or validate with the OK key. Esc jumps back up to the previous menu level, until getting back to the base screen (with level display).
  (details: page 46)

1.2. Audio settings

- With factory settings, the active inputs are the analog ones, and clipping level is set at +16 dBu for both inputs and outputs.
- Enter the menus using OK, select **Setup** then **Audio**. Perform necessary settings, *see details further on page 59*.

1.3. Select and set up network to be used: wired networks

- Select network: **Setup** menu then **Network**, then select **Change Network**, OK. In the proposed choice, select the desired network (**ISDN**, **LL**, **Ethernet**, **POTS**). Validate with OK.
- Select the audio coding: menu **Setup** / **Algorithm**, then **Other**, then OK. Browse the available choice with the arrows, and make a selection with OK. Restart the same procedure to change for another coding setup.
  ⓘ *The available choice depends on the transmission network! For more details on coding, see page 14.*

1.3.1. Set up an ISDN link

- If needed, select the protocol with **Setup** / **Network** / **ISDN Parameter** / **Protocol**. *More details: see page 53.*
- Enter the remote number to dial, using the keypad, and press the 📞 key.
- If more than one B channel is involved due to the coding algorithm used, you must enter a second number, then 📞, and so on. If the last dialled number is adequate, just confirm by pressing 📞 without typing a number again.
- Hang up with the 📵 key to release the connection (you must confirm by pressing 📵 again).

1.3.2. Set up a (wired) IP link

Over a public IP network, and especially when no SIP server is used, it is highly recommended to use a STUN server.
The address of a STUN server can be set in the SCOOP 5 html pages (*see page 90*) or via the menu: **Setup** / **Network** / **AoIP Parameter** / **STUN Server**, enter the address of a STUN server (*we propose our server* ***stun.aeta-audio.com****, look also the support pages on our web site* www.aeta-audio.com). Enable or disable STUN with **Setup** / **Network** / **AoIP Parameter** / **STUN Mode** (**On** or **Off**).
More details: see page 120.

- Check the Ethernet interface is active thanks to the LED on the Ethernet socket on the back, and check an IP address is allocated: menu **Tools** / **About** / **Local IP**.
- The default setting uses a DHCP server to get an IP address, which is suitable for most occasions. *In other situations, look for more details on page 36.*
- Using the keypad, enter the remote number to call (numeric IP address, or SIP URI if a SIP server is used), then press the key.
- Hang up with the key (you must confirm by pressing again).
- *When using a SIP server, some data must be entered beforehand using the* ***AoIP Parameter*** *menu; for more details, refer to page 51.*

1.3.3. Set up a POTS link

- For connecting the line, you must plug in ISDN socket #1 an adapter cable (included with the POTS option), which provides an RJ45 socket for the ISDN line on one hand, and on the other hand an RJ11 socket. Plug the POTS line in the latter.
- If needed, adjust the POTS line parameters using **Setup** / **Network** / **POTS Parameter**. *Details on these settings: see page 54.*
- Enter the remote number to dial, using the keypad, and press the key.
- Hang up with the key to release the connection (you must confirm by pressing again).

1.3.4. Set up a leased line (LL)

No connection procedure is involved with such link, as it is automatically set (or reset) as soon as the needed connections are done and the settings are correct on both sides of the link.

See in this manual on page 53 for configuration details.

1.4. Select and set up network to be used: mobile networks

To set links over a mobile network, an antenna connection is required (*more details on page 32*), and you must have a SIM card with a subscription suitable for the use. Specifically, for an IP mode transmission the subscription must include access to data transmission, and RTP audio streams must be allowed.

- While the unit is switched off, insert t he SIM card into the drawer on the rear side of the SCOOP 5.
- Switch on the SCOOP 5 (depress ⏻ a few seconds).
- Enter the menu **Setup** / **Network** / **Other (Network)** / **Mobile (Int.)** / **Mobile Parameter** / **PIN**. Enter the PIN code for the SIM card using the keypad then OK.

1.4.1. Set up a link in voice mode

This mode allows communicating with any telephone terminal through the regular telephone service. It also allows to benefit from the 7 kHz wide band service known as "HD Voice" whenever the remote terminal is compatible and the network supports the service.

- In the menu **Setup** / **Network** / **Mobile Parameter**, select **Mode** / **Cellphone**
- Afterwards, go to the menu **Setup** / **Network** / **Mobile Parameter** / **Network Settings** / **Preferred Technology** / **Auto**
- The **Network Select** menu enables you to select among the available operators, if your mobile subscription entiltes you to do so.
- Enter the remote number to dial, using the keypad, and press the ✆ key.
- Hang up with the ✆ key to release the connection (you must confirm by pressing ✆ again).

1.4.2. Set up a (mobile) IP link

⚠ Over a public IP network, and especially when no SIP server is used, it is highly recommended to use a STUN server.
The address of a STUN server can be set in the SCOOP 5 html pages (*see page 90*) or via the menu: **Setup** / **Network** / **AoIP Parameter** / **STUN Server**, enter the address of a STUN server (*we propose our server* ***stun.aeta-audio.com****, look also the support pages on our web site* www.aeta-audio.com). Enable or disable STUN with **Setup** / **Network** / **AoIP Parameter** / **STUN Mode** (**On** or **Off**).
More details: see page 120.

- In the menu **Setup** / **Network** / **Mobile Parameter**, select **Mode** / **IP Mode**
- Come back to the menu **Setup** / **Network** / **Mobile Parameter** / **Network Settings** / **Preferred Technology** / **Auto**
- Go to the menu **PS Settings** / **APN**: enter the operator's APN code, using the keypad, then press OK.
- Come back to the base screen using the Esc key.
- Using the keypad, enter the remote number to call (numeric IP address, or SIP URI if a SIP server is used), then press the ✆ key.
- Hang up with the ✆ key (you must confirm by pressing ✆ again).
- *When using a SIP server, some data must be entered beforehand using the* ***AoIP Parameter*** *menu; for more details, refer to page 51.*

2. Functions

The following synoptic diagram shows the basic functions of the equipment.

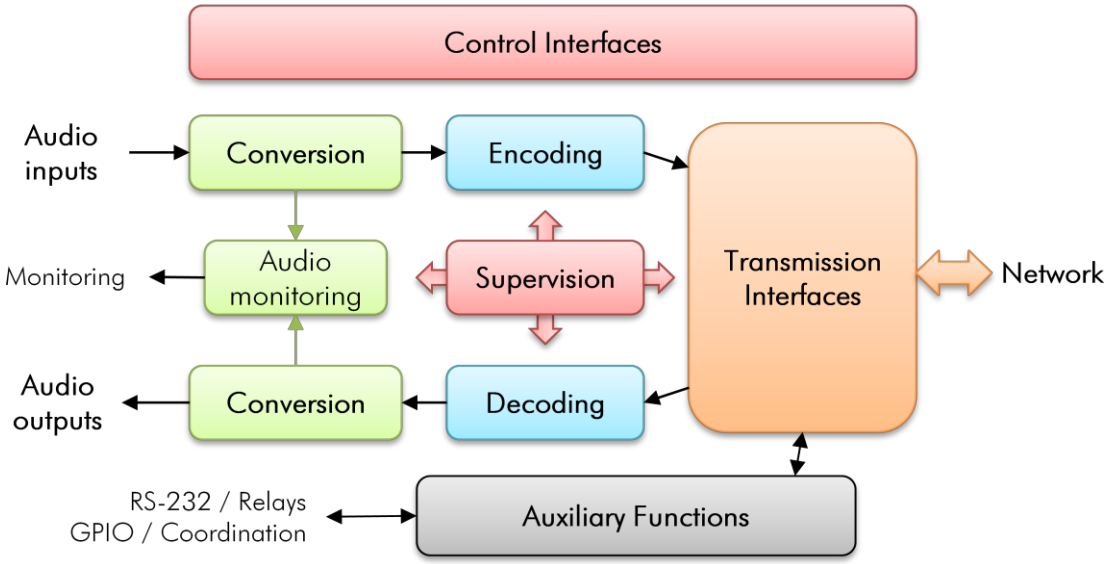


Figure 1 – Functional diagram of the equipment

The audio signals to be transmitted are converted (when needed) to digital format, then the encoding function reduces the bit rate, using a selectable algorithm; the resulting bit flow is sent to one of the available transmission interfaces: Ethernet interface, permanent link data interfaces (X21/X24/V35), ISDN interface(s), PSTN interface, mobile network...

The transmission interface functional block also extracts compressed data coming from the network and sends them to a decoding block that reproduces uncompressed audio data. Last, the audio signals are output to both digital and analogue outputs.

Monitoring the audio interfaces is possible thanks to a headphone and level meters for the inpts/outputs.

In addition to the main task of transmitting an audio programme, the SCOOP 5 can also transmit auxiliary information, usually by embedding them inside the transmitted audio streams.

Supervision and controlling the unit is performed using various remote control interfaces, and of course by means of the displays and controls on the front panel.

2.1. Transmission interfaces

The SCOOP 5 features in all versions an Ethernet interface for IP protocol networks, and leased line transmission interfaces (X24/V11).

One or two ISDN come in addition on some versions.

PSTN/POTS and mobile network access is optionally available.

2.1.1. Ethernet/IP interface

The IP interface is a 10BaseT/100BaseT Ethernet interface allowing transmission of the audio programmes in a wide range of possible bit rates. The audio stream is always transported under the RTP/UDP protocol.

IP unicast mode

The most classical transmission mode is unicast: audio connection with one distant device, generally bidirectional. This mode can be used on all types of networks links, LAN or WAN, including links via Internet. The SCOOP 5 implements the SIP protocol, which allows it to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Links can be set up in two ways:

- "Peer to peer" connection between two compatible units
- Use of a SIP proxy server to set up the link, or a SIP PBX

Details about the SIP protocols can be found in the annex (see 7.2, Overview of the SIP protocol).

The audio coding algorithm can be selected depending on the required quality and the available network bandwidth. The following algorithms are currently available:

Codec	Bit rate (coding)	Bit rate (total)[1]	Audio bandwidth	Typical use, main features
G711	64 kbit/s	86 kbit/s	3 kHz	Voice, telephony Compatible with IP phones
CELP	24 kbit/s	28,5 kbit/s	7 kHz	Suitable for high quality speech; Low network bandwidth consumption
G722	64 kbit/s	86 kbit/s	7 kHz	High quality speech. Compatible with some IP phones.
MPEG Layer II	64 to 256 kbit/s	73 to 275 kbit/s	Up to 20 kHz	Highest quality, suitable for speech and music
MPEG AAC-LC	16 to 256 kbit/s	30 to 277 kbit/s	Up to 20 kHz	Low bit rate, suitable for speech and music
MPEG HE-AAC and HE-AAC v2	16 to 128 kbit/s	23 to 139 kbit/s	Up to 20 kHz	Very low bit rate, suitable for speech and music
4SB ADPCM	128 or 256 kbit/s	173 or 301 kbit/s	15 kHz	Low latency, suitable for speech and music

Table 2 – Overview of algorithms available in IP mode

[1] Informative value; higher than the "net" encoded audio bit rate because of the protocol overhead

IP multicast mode

The multicast mode allows an encoder device to transmit an audio programme to several decoders by sending a single encoded stream to a multicast group address. The link is unidirectional by nature. This mode can be used on a local area network, and on larger private networks that can manage the multicast mode. On the other hand, Internet cannot support this routing mode.

In this mode, the SCOOP 5 uses the RTP protocol to manage the audio stream, like in the unicast mode, but the SIP protocol is not applicable here; instead a proprietary signalling system is used. As the link is unidirectional, the unit has to be set either as a "sender" in order to encode and transmit the audio stream to the selected group address, or as a "receiver" to receive and decode such stream coming from a "sender" device.

The audio coding algorithm can be selected with just the same capability as for the unicast mode described above.

SIP and SDP protocols

The SIP protocol is a signalling protocol, used for IP connections, which allows the SCOOP 5 to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Details about the SIP protocols can be found in the annex (refer to 7.2, Overview of the SIP protocol).

One significant advantage is the inclusion of SDP, a protocol which allows the connecting devices to automatically negotiate and agree on the coding profile to use. Thanks to this system, it is not necessary to set the units in the same way before setting up a connection. Moreover, the calling party needs not know how the remote unit is configured before initiating a link.

ⓘ *Note: the SIP protocol does not mandatorily imply the use of a server. Codecs can set up point-to-point links using this protocol, and benefit from some its advantages.*

Packet replication

SCOOP 5 also proposes an RTP transmission mode with enhanced reliability, using packet replication. When enabling this mode, every packet is transmitted twice; with such system a lost packet has no effect since the receiver still gets the other copy of the packet. In this way, stable links can be obtained even with a high packet loss rate. Of course, as a disadvantage the bit rate is double; you must make sure this stays compatible with the transmission medium.

Remote control via IP

In addition, the Ethernet interface can be used for configuring or remote controlling the unit, with two control methods:

- SCOOP 5 provides html pages which allow to get complete control over the unit using a web browser, via port 80 (default port for http protocol). See in chapter 5 the detailed operating mode.
- TCP port 6000 can be used for "command line" control, suitable for codec supervision software such as Codec Live, MDC.net, etc.

2.1.2. Leased line interfaces

For transmission over leased lines, the codec includes two X24/V11 ports which can run at 64 kbit/s, 128 kbit/s, 192 kbit/s, 256 kbit/s and 384 kbit/s bit rates.

Only one X24/V11 port is used, selectable among the two physical interfaces. The other, however, send the same data as the active one, and it can be used e.g. for a redundant link.

For "leased line" transmission, the codec synchronises on the network clock provided by the X24/V11 interface. If no suitable clock is present, the system folds back to an internal clock.

2.1.3. ISDN interfaces

For access to the ISDN, the transmission interface is one or two S0 BRI (Basic Rate Interface), for transmission over one to four 64 kbit/s B channels. Thus, the total available bit rate ranges from 64 to 256 kbit/s.

In "double codec" mode, only the first BRI is used, with each of the possible links using one B channel on this single line.

The codec synchronises itself onto the ISDN network clock when a link is active.

Network protocols

Available protocols:

- "Euro ISDN" (or ETSI), default protocol valid for a large number of countries, especially all over Europe.
- "NTT": valid for the Japanese network of NTT
- "NI-2": valid for numerous operators in North America. This choice is also suitable for the connection to network equipment with "NI-1" protocol.

In North America (USA and Canada), the available interface is often a U0 interface (instead of S0). In such case an "NT1" network adapter must be inserted between the line and the SCOOP 5. Such adapter can be found on the local market.

5A System®

Setting an ISDN connection is often difficult, at least because of the numerous coding parameters to be set. Moreover, with most proprietary algorithms, it is mandatory for the two devices to have exactly the same settings, otherwise the connection will fail, and sometimes it is not easy to find out the reason.

5A stands for Aeta Audio Advanced Automatic Adjustment. This system makes it easier to set an ISDN connection, because the codec, on receiving a call, automatically adjusts itself, following the calling party algorithm and parameters.

When the 5A System is enabled on the unit and a call is received, the unit first detects the coding algorithm used by the calling codec, and also senses its parameters: audio mode (mono, stereo…), sampling rate, bit rate, inverse multiplexing protocol, etc. Then the unit can decode the compressed audio from the remote unit. In addition, the unit will use these same settings for encoding and sending audio to the remote unit, so that the remote unit can also decode the outgoing audio programme. The whole process just takes a few seconds. Of course, all compatible coding configurations can be detected automatically by the 5A System.

Note that the 5A system is only active for ISDN connections.

J52

The ITU-T J52 recommendation was defined in order to allow the interoperability of multimedia terminals over the ISDN[1], using common coding standards. It includes the following features:

- Support of ITU-T recommended coding algorithms: G711, G722, MPEG Layer II
- Framing as per ITU-T H221 recommendation, ensuring byte synchronisation and interchannel synchronisation when more than one 64 kbit/s B channel is required for the desired bit rate;
- Interoperation procedures according to ITU-T H242 recommendation;
- In the case of MPEG encoding, optional protection against transmission errors (Reed-Solomon error correction codes).

Details about MPEG and J52 can be found in the annexes (refer to 7.1, Additional information on the algorithms and protocols used).

It must be noted that, thanks to the interoperation protocol, J52 codecs, when setting up a link, can negotiate automatically and agree on a configuration that is compatible with the capability of both units (regarding bit rate, channel mode, etc.). In this way, when the units differ in their capability (or make), the resulting configuration may be different from expected beforehand, but in most cases the link will work and audio will be transmitted.

As another useful consequence, this also gives users more tolerance to mistakes when configuring the units on the two sides of the transmission links, as the codecs will adapt automatically even with differences in the initial settings of the two units.

Symmetric or asymmetric coding

In most operating cases, the codec sets up **symmetrical** links, wherein the encoder and decoder use the same encoding/decoding algorithm with same settings (channel mode, etc.). In other words, the link is "full-duplex" and the same encoding/decoding type is used on both directions.

For a link over the ISDN, in fact there are cases when the link is **asymmetric**, with a different coding mode in each direction. This may happen in some cases with the J52 protocol. To give some examples, it is possible to send MPEG Layer II in one direction and receive G722 in the reverse direction, or send MPEG stereo and receive MPEG mono, etc.

2.1.4. POTS interface

The interface is a "two wire" analog telephone access, with characteristics adjustable depending on the country. Dialling normally uses DTMF, but for older switching equipment it is possible to use pulse dialling.

SCOOP 5 includes a V34 modem which transmits via this line a bidirectional audio flow, encoded at a nominal 24 kbit/s bit rate. Depending on the line quality and the quality of the link with the remote codec, this bit rate is automatically negotiated and dynamically adjusted from 12 to 24 kbit/s.

A "protected" mode can be activated, which increases the resilience to transmission errors, at the cost of a higher latency (encoder to decoder delay). *You must make sure to set this parameter the same way on both devices / both ends of the link.*

[1] J52 is only relevant for ISDN connections

2.1.5. Mobile network access

Units equipped with the "Wireless" option include an integrated module for access to 2G/3G/3G+ mobile networks, and a holder for a SIM card.

Depending on the version the accessible networks are 2G (GSM, EDGE), 3G (UMTS), 3G+ (HSDPA, HSUPA, HPA…), and 4G/LTE.

For the operation, at least one multiband antenna (to be selected for compliance with the mobile network characteristics) must be connected on SCOOP 5. An antenna "diversity" feature gives an opportunity to improve the stability when the radio reception quality is poor, by connecting a second antenna.

Lastly, SCOOP 5 can display the received SMS messages.

Mobile voice mode – HD Voice

The integrated module allows to use the mobile phone service, for communicating with all ISDN or PSTN telephone terminals or hybrids, or with other mobile terminals. The quality is in such case that of mobile connections, with a 300-3400 Hz bandwidth and coding such as GSM, EFR, AMR...

Now many mobile networks also propose "HD Voice", an extension of this mobile telephone service. With this new capability, compatible terminals implement the AMR-WB coding algorithm (standardised as G722.2 by the ITU-T) and provide speech transmission with a 50-7000 Hz bandwidth and a quality very similar to the well-known G722. Automatic fallback to the standard coding takes place if the network does not support the service or one of the terminals does not feature this capability.

No special subscription, other than to the regular telephone service, is needed, but for most operators only the 3G/3G+ base stations support the service.

ⓘ *This sometimes makes people believe that HD Voice is related to the mobile IP service, but this is definitely not the case.*

More and more mobile phones now support this service, especially (but not only, and not all) smartphones. All AETA codecs in "Wireless" version support HD Voice, namely:

- Scoopy+ HD (*except old units, in doubt consult us*)
- SCOOP 4+ in "wireless" version
- ScoopFone HD
- SCOOP 5

Mobile IP mode

The other service available with mobile access is the data packet transmission mode, abbreviated as "PS" (for Packet Switched), with IP protocol.

This mode brings similar capabilities as a wired IP connection via the Ethernet interface, as described above in 2.1.1, with some distinctive characteristics:

- This requires a subscription including access to the data service, with conditions compliant with the application. Among other requirements, an APN (Access Point Name) must be provided that allows this type of media stream.
- The available bit rate depends on various factors; first the network technology (2G/3G/3G+…), but also the traffic level in the radio cell, the operator's network capacity, possibly the type of subscription. This may bring on restrictions for the usable compression algorithms.
- The multicast mode is not available on mobile networks
- Setting a link implies first activating the data connection, before actually initiating an audio stream transmission link

Using an external USB module

Instead of the integrated module, it is possible to plug a USB mobile module or "key", in order to access mobile IP transmission, with more or less similar conditions as described above.

ⓘ *However be aware:*

- This capability is optional
- The USB module must be from the list of devices supported by AETA. *As this list is evolving, please check our web site for up to date information.*
- The "HD Voice" mode is not available in this way
- USB devices do not feature antenna diversity

2.1.6. Managing calls

Apart the LL mode (leased line) which is for a permanent connection, the audio transmission implies a link/session setup phase.

One of the transmission interfaces is selected as the *default interface* on the SCOOP 5.

A call towards a remote unit, initiated by the user of the SCOOP 5, is implicitly sent through this default interface.

On the other hand, an incoming call on any interface (regardless of the default interface) can be processed and the link established, under following conditions:

- The "called" interface must be connected and active. As an example, for mobile IP the data connection must be active.
- The codec must not be already busy with another connection.

If the call comes on an interface other than the default interface, the codec first switches to the suitable interface, and then processes the incoming call. When the link is released, it will come back to its previous state (and default interface).

2.1.7. Securing a link with a backup connection

When a fixed link (LL) is used, it is possible to use another network access in order to set a temporary backup link in case the nominal LL link fails. The unit will then switch to a backup mode (ISDN or IP depending on the selected backup arrangement), and provide the audio transmission via the backup network access. More precisely, on one end of the link the codec will switch to the backup mode and "call" its counterpart via the ISDN or and IP network. On the other end the unit will switch to the backup mode when it receives the call on its ISDN or IP interface. The operating mode and configuration for this backup feature are detailed further in following sections (3.5.4, Setting up a backup link).

2.2. Audio encoding and decoding

SCOOP 5 features a wide range of coding algorithms. Their availability depends on the transmisison network used. Besides, the MPEG family algorithms feature a large configuration flexibility.

The table below synthetically describes the capabilities with the various transmission media:

Codec	Audio channels	Frequency (kHz)				Bit rate (kbit/s)	Wired				Mobile		
		16	24	32	48		PSTN	LL	ISDN	Ethernet	3G+ LTE	UMTS	Voice
G711	Mono					64							
GSM, AMR	Mono					-							
AMR-WB	Mono					-							
CELP	Mono					12 → 22.8							
CELP	Mono					24							
G722	Mono					64							
G722-H221	Mono					64							
G722-H242	Mono					64							
TDAC	Mono					64							
4SB ADPCM	Mono					128							
4SB ADPCM	Stéréo					256			[2]				
MPEG L2	M / S					64							
MPEG L2	M / S					128							
MPEG L2	M / S					192			[2]				
MPEG L2	Stereo					256			[2]				
MPEG L2	Stereo					384							
AAC-LC	M / S					16 → 56							
AAC-LC	M / S					64							
AAC-LC	M / S					96							
AAC-LC	M / S					128							
AAC-LC	M / S					192							
AAC-LC	Stereo					256							
HE-AAC	M / S					16 → 56							
HE-AAC	M / S					64							
HE-AAC	Stereo					96							
HE-AAC	Stereo					128							
HE-AAC v2	Stereo					16 → 56							
HE-AAC v2	Stereo					64							

IP networks

Possible audio bandwidth:

3 kHz 7 kHz

15 kHz 20 kHz

[2] : requires two ISDN lines

Table 3 – Available coding depending on network

This table applies to the normal mode, single codec.

In double codec ISDN mode, only G711 and G722 algorithms are available for each codec.

The following chapters bring some precisions about the important features of the various algorithms and protocols available.

2.2.1. G711 coding

Application: telephony, coordination. Low latency.

G711 is the standard coding used for voice transmission on public telephone networks, and features 300 to 3400 Hz audio bandwidth. This algorithm is typically used for links over IP networks with IP telephones or VoIP gateways. Via ISDN, G711 is used for links with telephones or hybrid devices.

G711 is available only for IP or ISDN transmission

2.2.2. Mobile telephony coding: GSM , AMR

Application: telephony, mobile coordination. Moderate latency.

These algorithms are exclusively used for speech transmission over mobile telephone networks, with a 300 to 3400 Hz audio bandwidth. Gateways perform, whenever needed, transcoding in order to interface with fixed PSTN, ISDN and IP networks.

2.2.3. Mobile HD Voice coding: AMR-WB

Application: commentaries, mobile coordination. Moderate latency.

The AMR-WB coding (standardised as G722.2 by the ITU-T) is used between compatible mobile terminals, when the mobile network supports the so called "HD Voice" service, and provides speech transmission with a 50-7000 Hz bandwidth.

SCOOP 5 automatically implements this algorithm in mobile voice mode every time it is possible, and automatically falls back to standard voice coding if not (when network does not support, or the remote terminal is not compatible).

Unfortunately it is not possible to see directly whether AMR-WB is active or not at a given moment. You have to rely on your listening skills! However, it has to be active if the conditions are met: a) support from the network on both sides of the link, b) both terminals compatible, c) service continuity from end to end[1]

2.2.4. CELP coding

Application: commentaries, coordination. Low capacity transmission channels

This algorithm operates in mono for a net nominal bit rate of 24 kbit/s, and provides 7 kHz bandwidth and a quality close to G722 for a much lower bit rate.

CELP is available for IP or PSTN networks. For PSTN (POTS), the bit rate may be reduced to adapt for the line quality, among the following values: 12 kbit/s, 14.4 kbit/s, 16.8 kbit/s, 19.2 kbit/s, 21.6 kbit/s, 24 kbit/s. The bandwidth is reduced accordingly.

Still for POTS lines, in addition to the normal mode two "protected modes" are available, bringing an increased resistance to transmission problems, however with an increased latency as well.

[1] At the time of writing, this requires both units to be on the same network: same operator, same country

2.2.5. G722 coding

Application: commentaries, coordination. Low latency.

This mono coding algorithm at a 64 kbit/s bit rate is a reference for commentaries, and features a 50-7000 Hz bandwidth.

It is available for LL, ISDN, IP networks (wired or mobile).

For leasedlines or ISDN, three synchronisation modes are possible:

- "Statistical recovery" byte synchronisation method (alias SRT);
- H221 synchronisation; in this case, 1.6 kbit/s from the compressed data are used for this;
- H221 synchronisation and H242 protocol. This is only available for the ISDN mode.

H221 synchronisation is highly recommended when possible, as it features higher reliability and faster recovery time, while degradation (because of the bit rate used for framing) is minimal.

H242 protocol, the most flexible mode, is recommended by the ITU-T, and is included in J52. However, the mode with H221 synchronisation but without H242 protocol can be useful for compatibility with old generation codecs which did not use this protocol.

No specific synchronisation is required for the IP mode.

2.2.6. TDAC coding

Application: commentaries, mono music via ISDN with only one B channel.

The SCOOP 5 can also include the TDAC algorithm. TDAC is for Time Domain Aliasing Cancellation; this is a transform coding based on an MDCT (*Modified Discrete Cosine Transform*), encoding a 15 kHz bandwidth mono signal at a 64 kbit/s bit rate.

TDAC is available as an option, only for the ISDN mode.

2.2.7. 4SB ADPCM coding

Application: commentaries, mono or stereo music. Low latency.

4SB ADPCM operates either in mono at a 128 kbit/s rate, or in stereo at 256 kbit/s, for a 15 kHz bandwidth. It features a very low latency which makes it very interesting for live duplex transmission. It also has the advantage to be very little sensitive to tandem coding.

4SB ADPCM is available for LL, ISDN; wired or mobile IP. However it is not recommended for mobile networks, because it requires a high bit rate and it is highly sensitive to packet losses, which can be frequent on such networks.

2.2.8. MPEG Audio Layer 2 coding

Application: mono or stereo music, high quality.

As shown on Table 3, this coding algorithm features a maximum flexibility, with many variations for bit rate, mono or stereo channel mode, sampling rates...

The two channel modes exist in three variations:

- Stereo: coding of each channel stays independent
- Dual mono: coding is similar to the previous case, but this choice applies to channel with no acoustic relationship, e.g. two languages for commentaries
- Joint stereo: applies to stereo programme, but here the encoder exploits the interchannel correlation for improved coding. To be used only for a stereo programme

The 16 and 24 kHz sampling rates feature a moderate bandwidth (respectively 7 kHz and 10 kHz) and are rather useful for commentaries.

ⓘ *The latency is rather high with these sampling rates*

MPEG L2 is available for LL, ISDN; wired or mobile IP networks.

ISDN mode specific aspects: J52

For ISDN transmission, MPEG L2 is proposed with two variations:

- "MPEG L2 J52" variation, using the ITU-T J52 protocol for the link negotiation and inverse multiplexing (B channel aggregation for connections requiring more than 64 kbit/s). Moreover, optional protection against transmission errors (FEC) is also available. For more details refer to 2.1.3, ISDN interfaces (page 9).
- MPEG L2" variation, without the J52 protocol, using a system for inverse multiplexing two B channels which is proprietary but compatible with several codecs on the market. The bit rate is limited to 128 kbit/s with this option.

LL mode specific aspects

In leased line mode, an optional protection against transmission errors (FEC) is also available. This is the same FEC system as implemented for the J52 variation of the ISDN mode.

2.2.9. MPEG AAC algorithms

Application: mono or stereo music, Low capacity transmission channels.

These (optional) algorithms feature a very high compression ratio, for a given audio quality, compared to Layer 2. They can operate at a sampling rate of 32 or 48 kHz, and several bit rates: 16, 20, 24, 32, 40, 48, 56, 64, 96, 128, 192, 256 kbit/s. Three coding variations are available:

- MPEG AAC-LC ("Low Complexity"): lower compression than other variations, but lower latency.
- MPEG HE-AAC ("High Efficiency" AAC): higher compression, and the bit rate is limited to 128 kbit/s for this variation.
- MPEG HE-AAC v2 ("High Efficiency" AAC version 2): compared to the above, this coding further enhances the performance for a stereo program (not available for mono). The bit rate is limited to 128 kbit/s for this variation.

AAC codecs are available as an option for ISDN, and wired or mobile IP networks.

2.3. Audio interfaces

2.3.1. Analog interfaces

The analogue inputs and outputs are balanced, and the input and output gains are adjustable.

The input to the encoder is selectable between the digital audio input and the stereo analogue input. The output from the decoder is always provided on the stereo analogue output.

The sampling frequency of the analogue ⇔ digital converters is automatically set depending on the coding algorithm used for transmission.

2.3.2. Digital interfaces

The equipment also provides digital audio inputs/outputs in AES/EBU format.

The input to the encoder is selectable between the digital audio input and the stereo analogue input. The output from the decoder is always provided both on the digital output and the stereo analogue output.

The digital audio interfaces are usually locked to the digital audio input ("genlock" mode), but alternatively they can be synchronised to the internal clock reference of the codec (mode called "Master").

Sampling rate conversion is automatically performed, whenever needed, depending on the coding algorithm used for transmission.

ⓘ *As an important consequence, the selection or value of the sampling frequency of the AES/EBU input/output is* ***completely independent*** *of the sampling frequency of the compression algorithm.*

One should also be aware that the various audio settings have no relationship with those for the other side of the link (the remote codec), whatever they are: selection of analog or digital source, sampling, rate of AES interfaces, etc. This configuration is only relevant for the local installation.

2.3.3. Audio monitoring

Audio programmes can be monitored of the audio input (before encoding) and at the audio output (after decoding the received signal).

First, the programme level is displayed on the graphic screen; both input and output levels are monitored simultaneously.

Besides, a test output on a stereo headphone jack allows monitoring either the encoder or decoder audio signals.

ⓘ *Note: as the audio output is monitored immediately after decoding, this monitoring position is not sensitive to the possible activation of the audio test loop (see further the chapter about test loops), contrarily to the physical audio outputs (both analog and digital).*

2.4. Auxiliary functions

The main function of the SCOOP 5 is the transmission of one or two main audio programmes, but it also provides auxiliary functions for transmitting data or additional signals, inside the same stream (or, more generally, the same session).

These features are only compatible with AETA products, because they are not inside the scope of independent standards.

The availability of these functions depends on the coding algorithms, and on the transmission network. The following tables show these capabilities for the various networks.

ⓘ *No auxiliary function is available for mobile voice transmission. Besides, for PSTN/POTS only "relay transmission" (2 loops) is possible.*

Codec	Relays		Data (bauds)					Coordination channel
	Isolated	GPIO	300	1200	2400	4800	9600	
G711								
CELP								
G722-SRT								
G722-H221								
G722-H242								
TDAC								
4SB ADPCM	X		X					X, Option
MPEG L2								Option
MPEG L2/J52								
AAC-LC								
HE-AAC								
HE-AAC v2								

X = exclusive (only one function at a time)

Table 4 – Auxiliary functions: over leased lines

Codec	Relays		Data (bauds)					Coordination channel
	Isolated	GPIO	300	1200	2400	4800	9600	
G711								
CELP								
G722-SRT								
G722-H221								
G722-H242	X		X					X, Option
TDAC								
4SB ADPCM	X		X					X, Option
MPEG L2								
MPEG L2/J52								Option
AAC-LC								
HE-AAC								
HE-AAC v2								

X = exclusive (only one function at a time)

Table 5 – Auxiliary functions: over ISDN

Codec	Relays		Data (bauds)					Coordination channel
	Isolated	GPIO	300	1200	2400	4800	9600	
G711								
CELP								
G722								
G722-H221								
G722-H242								
TDAC								
4SB ADPCM								
MPEG L2								
MPEG L2/J52								
AAC-LC								
HE-AAC								
HE-AAC v2								

Table 6 – Auxiliary functions: over IP networks (wired or mobile)

2.4.1. Transmission of isolated relays

When this function is activated, the codec transmits to the remote unit the status of two isolated current loops. The remote unit then opens or closes relay contacts according to the transmitted status. Conversely, as the function is bi-directional, the codec activates its two relays ("dry" isolated contacts) depending on the status of the two current loops on the remote unit.

For transmission over IP, this feature is always available whatever the coding algorithm. For the other networks, availability depends on the algorithm: see above tables.

With G722 or 4SB ADPCM, relay transmission cannot be used at the same time as another auxiliary function (see tables).

A typical application is the transmission of "on air" signals; a contact closure can be used for instance to light up an indicator or switch on other equipment.

2.4.2. Transmission of GPIO

For IP transmission, SCOOP 5 also allows, in the same way as the two isolated relays, to transmit 6 additional binary signals, routed to "GPIO" interfaces, which are not isolated. This feature is available whatever the coding algorithm.

2.4.3. Data channel

This function is not available in IP transmission mode.

In leased line mode or ISDN mode, a bi-directional data channel can be transmitted along with the compressed audio signals, by reserving a fraction of the transmitted bit rate. The equipment includes a serial asynchronous port for this purpose. The data are transparently transmitted end-to-end; hardware signalling is not available.

Availability depends on the coding algorithm, as well as the max possible baud rate: see above tables. With G722 or 4SB ADPCM, this data channel cannot be used at the same time as another auxiliary function (see tables).

A typical application is the transmission of data related to the RDS.

2.4.4. Coordination channel

This function is not available in IP transmission mode.

This function is available as an option. It enables the transmission of an auxiliary audio channel (or coordination channel), along with the compressed audio, by reserving 8 kbit/s from the transmitted bit rate. This channel uses a compression algorithm of CELP-HLTP type and provides a "voice grade" channel (300-3400 Hz pass-band).

This feature is only usable for some algorithms, on leased lines or ISDN: see above tables.

With G722 or ADPCM, the coordination channel cannot be used along with other auxiliary functions (i.e. data channel or relay transmission).

On a SCOOP 5 with this option, one of the sockets on the rear panel of the device includes one isolated balanced input and one isolated balanced output. An adapter cable, included with the option, provides an easy connection on XLR plugs.

2.5. Supervision and control interface

The control and supervision of the equipment (configuration, communication management, status monitoring), is carried out either "locally" thanks to a keyboard, an alphanumeric display, LED indicators, or using various remote control interfaces.

2.5.1. "Local" control

For local management, the front panel includes a large keypad, a backlit graphic LCD, and various LED indicators for essential status information.

2.5.2. Embedded html server: "web pages"

SCOOP 5 provides html pages that enable full control using a web browser, via port 80 (default port for the HTTP protocol). See on chapter 5 the detailed operation mode.

This control mode can be used from any computer regardless of its OS (or a mobile device with a web browser), and the embedded pages are compatible with all current browsers. No software installation is needed on the control device.

2.5.3. Remote control via a serial port or TCP/IP

Another remote control mode makes use of a dedicated protocol (AARC), via a TCP/IP connection on TCP port 6000. This "command line" mode is used by codec pool management systems such as Codec Live, MDC.net, etc.

This interface can also be operated using AETA's TeleScoop software.

Instead of TCP/IP over the Ethernet interface, it is possible to use the RS-232 serial remote control port available on the unit.

2.5.4. Additional Ethernet/IP interface

It is possible to get a second Ethernet/IP interface by plugging a USB/Ethernet adapter[1] on the USB socket. This interface can be used as a remote control port (html pages via port 80, or "command line" mode via port 6000), as an alternative to the normal integrated interface. *It should not be used for other functions (audio over IP).*

2.5.5. Loop control and status

In addition, the equipment features a "Loop control" function: call set up and release can be remote controlled with current loops and relays, instead of using for this the keyboard and/or the remote control port. Relay contacts provide a status (idle/on line) of the unit.

2.5.6. Alarm contacts

Besides configuring the equipment operating mode, this module supervises its status (detection of alarm conditions). On detecting operation or transmission faults, the equipment switches on indicators and relay contacts. Two alarm classes are defined:

- "Internal" alarm; corresponds to a major fault internal to the equipment;
- "External" alarm; corresponds to a fault whose origin is deemed external to the equipment (for example, transmission fault);

[1] Reference : Logilink UA0144, or other device shown on our web site support pages.

2.5.7. Configuration and dialling memories

To ease the operation, it is possible to store configuration memories, called "profiles". These belong to three categories:

- "Remote profiles", including the parameters for calling a given destination: dial numbers, coding algorithm, etc.
- "Local profiles" which memorise the network access characteristics. Recalling a local profile is a quick way to recover the configuration needed for connecting on a given line/network.
- "Snapshots" which memorise all the settings for the audio interfaces.

These various profiles can be used locally and also through the web pages, and they can be imported/exported from/to a computer.

2.5.8. Test functions

For maintenance purposes, some test loops can be activated. The following drawing schematically shows these test loops:

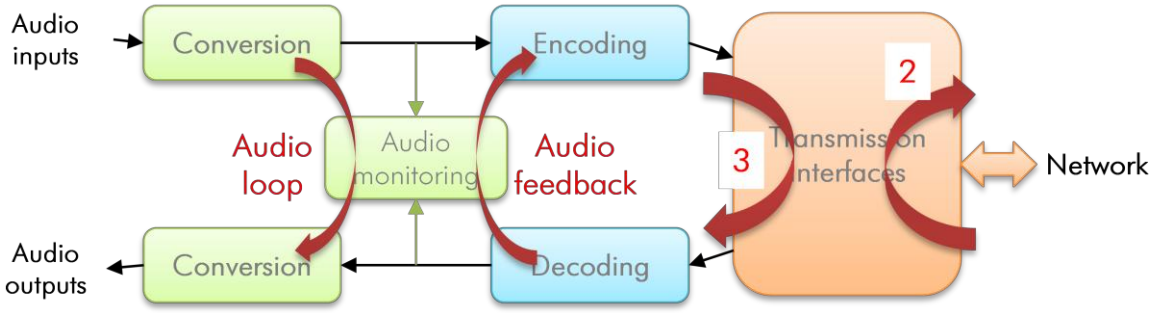


Figure 2 – Test loops

- "Audio loop": uncompressed audio data are looped from the input of the encoder to the input of the output conversion functional block. This loop redirects the audio input to the audio outputs;
- "Loop 3", or "Codec" loop: compressed audio data are looped just before the network interface;
- "Loop 2", or "Network" loop: this loop sends the received data back to the network; for the remote codec, the effect is the same as a loop 3 when the transmission works correctly;
- "Audio feedback" loop (audio output to audio input); this allows the codec to send back to the remote codec the signal it receives, after decoding and re-encoding.

3. Operation

3.1. General principles – Control means

The equipment control and supervision (configuration, status monitoring) is possible either in "Local" mode (front panel keypad and display, status indicators), or "Remote control" mode, thanks to an asynchronous serial port or an Ethernet interface.

As a general rule, the configuration parameters are saved in non-volatile memory, and restored when the unit is powered on.

Local mode operation is described in detail in chapter 4 (Detailed operating mode).

Remote control operation using a computer and a web browser, thanks to the embedded HTML server, is detailed in chapter 5: Operating mode – Embedded HTML .

SCOOP 5 can also be controlled by the optional TeleScoop software (installed on a Windows PC), which can control the other AETA codecs of the Hifiscoop, Scoop 3, Hifiscoop 3 and SCOOP 4+ ranges. Details about this supervision software can be found in the documentation and user manual of the TeleScoop software.

The SCOOP 5 can be remote controlled by third-party codec management software and systems. Please consult us for more information on the available offer in this field.

For controlling connections, it is also possible to use the "Loop control" function. When this special connection mode is selected, one can trigger a call by activating an input current loop (optically isolated), and release the line by de-activating this loop. In such case, an outgoing connection is established or released only by this way, and no more from the front panel or the remote control interface (however, all other parameters are still controlled from these interfaces as in the normal mode).

If "loop control" is not activated, it is always possible to use the loop to release a running connection (a pulse on the loop will release the line).

The loop control interfaces are described in 3.5.3 and 6.1.14.

Besides, whatever the connection mode (normal or loop control), a "dry loop" is closed when a connection is active.

3.2. Physical description of the equipment

The SCOOP 5 codec is housed in a 19 inches chassis of 1U height (44 mm or 1.75"). It includes a universal mains power supply. There is an option for powering from a 12V DC source (which can be used in parallel with the mains input, with priority to the latter).

3.2.1. Front panel

All the elements needed for local control are located on the front panel (see picture on page 26 below). This panel can be roughly divided in two areas:

On the left-hand side, one can find several LED indicators, an LCD and the main navigation keys. On the right-hand side are located the dialling keys and the call management keys. Lastly, at the extreme right side one can find audio monitoring elements.

On/Off switch and standby

First, completely on the left is located the ⏻ on/off key, and just besides the standby indicator (blue LED). When the unit is powered but in standby (blue LED on), keep the key pressed for at least 3 seconds to switch on the unit. When it is operating, keep the key pressed for at least 3 seconds to switch it off.

In addition to this manual "on/standby" switching, the unit automatically switches on when power is applied on its mains socket (or its DC input for those units equipped with the DC option).

The standby LED next to the ⏻ key is off in operation, but lights on when the unit is in standby with power applied.

Status LED indicators

The LEDs have the following meaning:

Marking	Color	Function
Alarm	*Red*	Shows an alarm
Sync 1	*Green/Red*	Off when unit is idle; *Green* when the decoder is synchronised on "Line 1"; *Red* if interface n°1 is active/connected but not synchronised or in error
Sync 2	*Green/Red*	Off when unit is idle; *Green* when the decoder is synchronised on "Line 2"; *Red* if interface n°2 is active/connected but not synchronised or in error
Line 1	*Green*	On when interface n°1 is active / connected
Line 2	*Green*	On when interface n°2 is active / connected

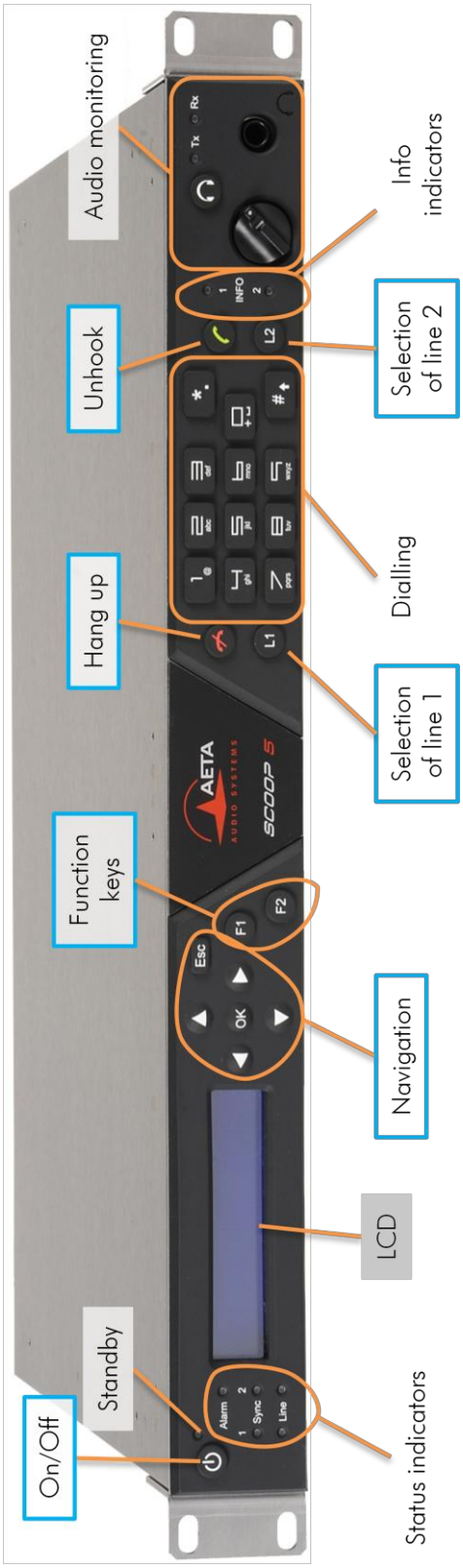


Figure 3 – SCOOP 5 front panel

LCD and navigation keys

Besides the backlit graphic LCD one can find the keys for navigating through the menus:

Key	Function
OK	Confirm a selection or enter data. From the base screen: switch to the root menu.
Esc	Escape to upper menu level; From the base screen: switch to the root menu.
Navigation keys	Keys used to move the cursor or browse through menu options. The ▶ can be used to enter a sub-menu (like the OK key). When entering data/numbers, the ◀ erases the character on the left of the cursor.

More details on the navigation and sub-menus can be found in chapter 4, dealing with detailed operating modes.

Function keys

The assignment of these **F1** and **F2** keys is programmable.

Call management and dialling keys

Key	Function
"Hang up"	Release an established link. *(except LL links, permanent by nature)*
"Unhook"	Start a link or accept an incoming call. *(except LL links, permanent by nature)*
Dialling keys **0 .. 9** ***.** **#⇑**	Keys for entering numbers or URI for the destination to call. These keys are also used to enter texts such as profile names, etc. Repeatedly press a key to get the characters other than the main figure. Keys **1** and "***.**" also allow entering special additional characters (not all marked on the keypad). The "**#⇑**" key is used for switching between numbers, lower case letters or capital letters. An indicator on the screen recalls the current type of characters.
L1	**In double ISDN codec mode**, this key is used to select line 1 for calling or hanging up. *Useless in normal single codec mode.*
L2	**In double ISDN codec mode**, this key is used to select line 2 for calling or hanging up.

"INFO" indicators

These LEDs show the state of the *received* information when the auxiliary function "Relay transmission" is active:

Marking	Color	Function
INFO 1	*Amber*	Shows the state of the "relay contact" n°1
INFO 2	*Amber*	Shows the state of the "relay contact" n°2

Audio monitoring

The audio signals can be monitored with a low impedance headphone connected on the front panel (1/4" or 6.35 mm stereo jack). The headphone volume is adjustable thanks to a potentiometer.

By pressing key , you can select to listen either the transmitted program (**Tx** indicator) or received audio (**Rx** indicator). The transmitted program is picked just before the encoder, the received program is picked just after the decoder.

Actions dealing with this area (connecting or disconnecting the jack, Tx/Rx selection, volume adjustment) never affect the transmitted or received signals.

3.2.2. Rear panel

All connections are done on the rear panel of the codec. The characteristics of the interfaces and layout of the sockets are detailed in chapter 6.1, Characteristics of interfaces.

The following elements are available (*numbers such as [17] refer to the following Figure 4 – Rear panel*):

Mains power socket [17]

This is an IEC type power socket. The unit starts up as soon as power is applied.

See details in 6.1.17 and 6.4 (Power supply).

DC 12V power socket [16]

This 4 pin male XLR is optional. *See details in 6.1.18, DC power supply (option).*

Audio inputs/outputs

- Analog inputs/outputs [9]: at the input, plug the audio cables into the female XLR sockets. At the output, plug the audio cables into the male XLR sockets. In mono mode, only "A" channel is used.
- Digital inputs/outputs [1]: a digital input (mono or stereo) in AES/EBU format (or SPDIF) can be connected on the female XLR socket, and a digital output in AES/EBU format is available on a male XLR socket.
- It is possible to select which input (analog inputs or digital input) is fed to the encoder for transmission. On the receiving side, the decoded signals are output both on the analog and digital outputs.

Ethernet interface [13]

This socket is a 100BaseT/10BaseT port, used for audio transmission over IP and/or for remote controlling the unit. This RJ45 socket is devised for a normal "straight" cable to an Ethernet hub or switch. The two integrated LEDs show the presence and activity of the network (green LED) and the interface mode: half-duplex (yellow LED off) or full-duplex (yellow LED on).

The configuration of the interface is described in 3.4, Initial setup of the Ethernet interface.

ISDN 1 and POTS socket (marked "ISDN 1/POTS") [5]

This RJ45 socket allows the connection to the ISDN, for the product versions which include this capability. The wiring is for connecting an S0 (BRI) ISDN line using a standard RJ45 cable.

Only this socket has to be used when only one line is needed (links using one or two B channels), which includes the operation in double ISDN codec mode.

For the units equipped with the PSTN/POTS option, this socket is also used to connect the POTS line. For this you can use the adapter cable provided with the POTS option: this adapter splits on two sockets the ISDN 1 line and the POTS line:

- RJ45 socket to get the ISDN 1 access;
- RJ11 socket to connect the POTS line: you must plug here a cable from the POTS wall socket[1] and terminated with a RJ11 plug.

Refer to 6.1.8 for more details on the pinout for the "ISDN 1" socket and the adapter cable.

[1] Whose type depends on the country and the building wiring

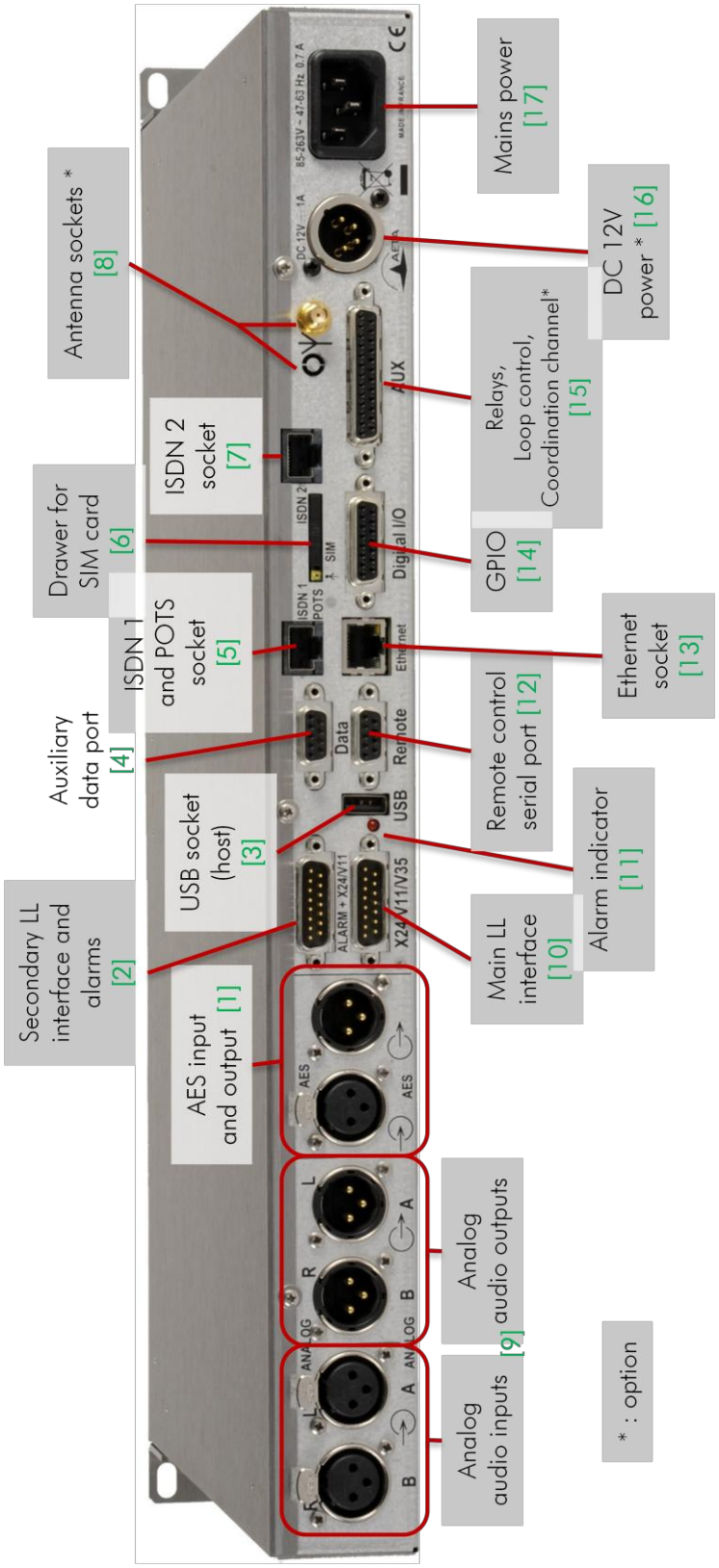

Mains power [17]
DC 12V power * [16]
Relays, Loop control, Coordination channel* [15]
GPIO [14]
Ethernet socket [13]
Remote control serial port [12]
Alarm indicator [11]
Main LL interface [10]
Analog audio outputs [9]
Analog audio inputs
* : option
Antenna sockets * [8]
ISDN 2 socket [7]
Drawer for SIM card [6]
ISDN 1 and POTS socket [5]
Auxiliary data port [4]
USB socket (host) [3]
Secondary LL interface and alarms [2]
AES input and output [1]

Figure 4 – Rear panel

ISDN 2 socket (marked "ISDN 2") [7]

This RJ45 socket allows the connection to the ISDN, for the product versions (TRIO 4B) which include this capability. The wiring is for connecting an S0 (BRI) ISDN line using a standard RJ45 cable.

This socket is not used when only one line is needed (links using one or two B channels), which includes the operation in double ISDN codec mode.

Main LL interface (marked "X24/V11/V35") [10]

This socket is used for the connection to data transmission equipment in the "leased line" mode.

The connector is 15-point male, Sub-D type. It is possible, if needed, to select the other LL port.

Details on this interface: refer to 6.1.5 - Ethernet.

Secondary LL interface and alarms (marked "ALARM + X24/V11") [2]

The 15-point male, Sub-D type connector includes two "form C" relays, signalling alarm conditions:

- "Internal alarm" contact;
- "External alarm" contact.

This port can be used as an alternative interface for connecting the codec to transmission equipment in "leased line" mode.

See the wiring in: 6.1.7 - "Alarm + X24/X21" interface.

Alarm indicator [11]

A red LED indicator also indicates that an alarm relay is activated. In the factory setup, every alarm cause sets the LED on, but by setting jumpers on the motherboard it is possible to program the indicator to react to only one type (internal or external alarm).

The pin-out of the socket and the detailed characteristics of the alarm relays can be found in chapter 6.1.7: "Alarm + X24/X21" interface (p. 102).

Drawer for SIM card [6]

In the units fitted with the mobile network access option, this drawer houses the SIM card enabling the access to the network and services.

ⓘ *The SIM card must be inserted while the unit is off (or in standby).*

To remove the drawer and the card, push the small button shown by the little arrow besides the drawer (*see picture below*).

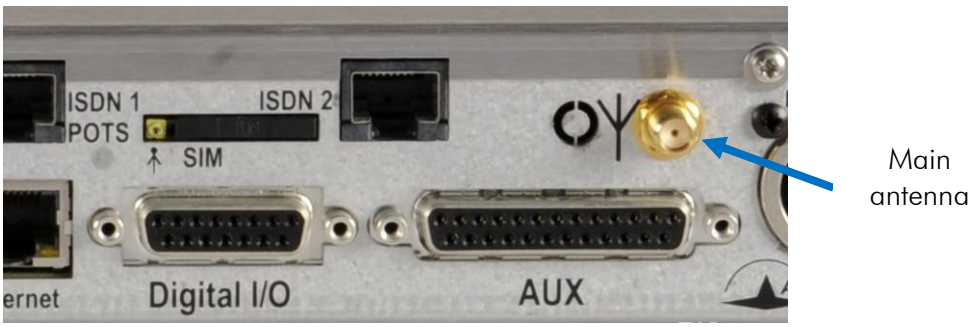


Figure 5 – Rear panel, SIM card

To set the SIM card in place, first insert it in the cavity inside the drawer. Make sure to have the cut corner on the right place, with the SIM chip visible. Then insert the whole assembly into the slot, with the SIM card chip facing down. Check that the drawer is well in its guides before pushing it completely in place.

ⓘ *If you have to use a smaller format µSIM card, you can use a SIM/µSIM adapter (available on request from AETA): first set the µSIM inside this adapter, afterwards use this assembly like a regular SIM card.*

Antenna sockets [8]

On the products fitted with the mobile network access option, these SMA sockets allow to connect one or two antennas (a multiband antenna is included with the mobile network option).

At least one antenna must be plugged on the main socket, the one that is shown on the picture above (the outmost socket, left side of the unit).

A second antenna is optional, but it allows to improve the reception quality in less favourable areas; it must be activated (configuration menus) if one is connected.

The antennas must cover the band(s) used for the operator and network services. Refer to the operator in doubt. The provided antenna covers the 900 MHz, 1800 MHz, 1900 MHz, 2100 MHz bands. It is compatible with almost all the 2G and 3G/3G+ networks in Europe.

USB socket [3]

This "host" socket allows the connection of a peripheral device, e.g. a mobile access USB module or "key" in order to access mobile IP transmission.

Refer to chapter 2.1.5 about this function.

As another use, an Ethernet adapter can be plugged for providing a secondary Ethernet/IP remote control interface; *see chapter 2.5.4 for this topic.*

Remote control serial port (Remote) [12]

This 9-pin female sub-D socket is an asynchronous serial interface port, usable for remote controlling the equipment thanks to a control and supervision PC.

For this capability refer to 2.5.3 Remote control via a serial port or TCP/IP.

Auxiliary data serial port (Data) [4]

This 9-pin female sub-D socket is an asynchronous serial interface port, usable for transmission of a bi-directional data channel (refer to 2.4.3 above, Data channel).

"AUX" socket [15]

This 25-pin female sub-D socket groups the interfaces for the relay transmission function (described in 2.4.1) and the (optional) coordination audio channel (optional, cf. 2.4.4).

It also includes loop interfaces for the loop control function (cf. 2.5.5), as well as a +5 V power supply that can be used to provide current for the loop and relay interfaces.

"Digital I/O" socket [14]

This 15-pin male sub-D socket is the interface for the GPIO transmission function (as described in 2.4.2).

Its wiring is described in: 6.1.16 - "Digital I/O" interface .

3.3. Installation and set up

3.3.1. Mounting and connections

Natural convection cools the equipment. Avoid obstructing the openings on the flanges.

To operate the codec, the minimum necessary connections to set up are (see details in the rear panel description):

- Power supply (mains and/or DC);
- Audio inputs and outputs (XLR sockets);
- Network interface: depending on the networks used, Ethernet interface, ISDN line(s), POTS line, X24/V11/V35 interface, or antenna(s) for mobile network access[1];

Whenever needed, the "ALARM + X24/V11" socket must be connected to an external supervision system (alarm relay contacts).

The pin out of the connectors is indicated in chapter 6.1: Characteristics of interfaces.

3.3.2. Initial set up

Before the first use, the equipment must be configured according to the desired operation mode: audio input/output format, local conditions (network interface parameters…). Then to set up links you must select the coding type and parameters.

For using the configuration menus, a password may have to be entered. After factory setting or after a complete configuration erasure, the password is "blank" (no password) and the access is unrestricted. Afterwards, if needed, a password can be programmed by the user and the restricted access mode can be activated.

For more details about the codec configuration, see chapter 4 - Detailed operating mode. The setup for the Ethernet interface is described in 3.4 below (Initial setup of the Ethernet interface).

3.3.3. Notes about the use of AES/EBU interfaces

When using digital audio interfaces, it must be decided whether the codec is "*master*" or "slave" regarding audio sampling clock synchronisation. In the first case, the codec derives the sampling clock from the network clock or an internal source, and the device(s) connected to the codec must synchronise to the same clock source.

The most common choice is rather the "slave" mode, to be used when it is not possible (or not desired) to synchronise the external equipment onto the clock of the transmission link or the codec. In this case, the AES/EBU interfaces should be set in the so-called "*genlock*" mode. When in this mode, the codec derives the sampling clock of the digital audio interfaces from its AES input (in other words is "gen-locked" onto the incoming AES signal), and sampling rate conversion (SRC) is used for interfacing to the coding parts.

ⓘ *It is mandatory in such situation to provide the codec input with an AES signal featuring the same sampling frequency as the external equipment, even if the codec is used only as a decoder.*

If this requirement is ignored, the unit will actually fall back to "master" mode. In such situation, clicks in the audio programme might be heard, especially when the resulting sampling rate is very different from that of the external device.

[1] In such case, a SIM card should be set as well in order to enable the mobile services.

If, on the contrary, it is decided to synchronise the external equipment (at 32, 48 or 96 kHz) onto the transmission clock of the network interface, the codec must be configured in "master" mode. In this case, the output is locked onto this clock, and it can be used as a reference to synchronise the equipment connected to the codec output. The digital audio signal at the codec input must then come from a device synchronised by this way.

When you do not use the digital audio interfaces, the "master" or "slave" mode has no effect on the actual operation. However, it is recommended in such case to select the "master" mode to avoid undesired alarms. Otherwise, with the "genlock" setting (which is the default factory setting), an alarm is raised because of the lack of a suitable signal on the AES input. In the "master" mode, the device ignores this error condition.

3.4. Initial setup of the Ethernet interface

The SCOOP 5 includes a 100BaseT / 10BaseT Ethernet interface, and the audio transmission can take place over an IP network through this interface. In addition, it is always possible to use the Ethernet interface to access the embedded html server or for remote control the unit via a TCP/IP connection (TCP port 6000).

An initial set up is needed for using one of these features of the Ethernet interface. For setting into operation, first connect the Ethernet interface to the network, using CAT5 wiring.

- Connection to 10BaseT or 100BaseT interfaces are both suitable, as the SCOOP 5 automatically switches to the adequate 10 Mbit/s or 100 Mbit/s mode.
- "Straightforward" patch cables should be used for a connection to a hub or a switch. Conversely, a "crossed" cable might be needed for special configurations (e.g. a test connection to a PC).

As a very first step, the Ethernet interface must be assigned an IP address, and related parameters. This phase is very simple when a DHCP server is available in the network. The menu to use is reached by **Setup** / **Network** / **Ethernet Parameter**.

3.4.1. DHCP server available

This is the simplest case, because the server will allocate a suitable IP address and give the unit the right settings. Select "**DHCP**" in the menu (**Setup** / **Network** / **Ethernet Parameter / Mode**). The unit will then automatically find the DHCP server and automatically set the parameters. You can read the IP address (allocated to the unit by the DHCP server) in the "About" menu (**Tools / About**).

Note that, as an additional advantage with DHCP, you do not need to change this setting later, even if you move the codec to another network, as long as it is still connected to a DHCP server.

3.4.2. "Static" IP configuration

When there is no DHCP server, you have to enter the settings manually. The IP address must be "available", i.e. not already assigned to other equipment. Ask support from the network administrator(s) as needed.

First select the manual mode, menu **Setup** / **Network** / **Ethernet Parameter** / **Mode** / **Manual.** Then, still in the menu **Setup** / **Network** / **Ethernet Parameter,** you must enter the following parameters:

Parameter	Notes
Local IP	Must be unique on the network
Subnet Mask	A typical value is 255.255.255.0
Gateway	
DNS Server 1	Domain Name Server (main)
DNS Server 2	Domain Name Server (secondary)

All addresses are in "dot-decimal" format, such as e.g.: 192.168.0.12, 10.0.54.123.

ⓘ *Note: in contrast to the configuration with DHCP, the "static" setting has to be reviewed each time you move the unit to a new physical site/network, as the previous IP addressing is probably not valid for the new location.*

3.4.3. Checking the IP configuration

The above configuration is kept in the unit's memory, and reloaded at each start.

To check the setting, you can read the IP address in the "About" menu (**Tools / About / Local IP**).

You can then also check that the unit is seen on the network and at the right address: from a computer connected to the same network, enter (in the command mode, or console mode depending on the OS) "ping *ipaddr*", where *ipaddr* is the IP address of the SCOOP 5.

If the response is positive, then you can proceed with the rest.

3.4.4. Optional configuration of the "link" Ethernet layer

With factory settings, the Ethernet interface is set for an auto negotiation of the "link" mode: speed (10 or 100 Mbit/s) and half-duplex or full-duplex.

This setting is suitable for almost all situations and usually you don't need to change it.

However, in the rare cases where this is needed, it is possible to set the desired mode manually: use the menu **Setup** / **Network** / **Ethernet Parameter** / **Link Mode** . The available choices are:

- **Auto-negotiation** (*standard setting*)
- **100BaseT, full-duplex**
- **100BaseT, half-duplex**
- **10BaseT, full-duplex**
- **10BaseT, half-duplex**

Whenever this setting is needed, it must be done before the IP configuration described above.

3.4.5. Configuration of the secondary Ethernet interface

In case an external USB/Ethernet adapter is used to provide an additional interface (see 2.5.4), an initial setup is also required for this interface. The procedure is similar to that for the main integrated Ethernet interface:

- First plug the adapter on the USB Interface.
- Connect the Ethernet interface to the network, using CAT5 wiring. *Normally the adapter's LED should blink depending on the network activity*. The additional interface is designated "**Ethernet 1**" in the unit's menus.
- Apply the same steps as described above (3.4.1 to 3.4.4), the settings are in this case accessed via the sub-menu **Config** / **Network** / **Ethernet 1 Parameter**.
- To check the IP address and status of the interface, go to the menu **Config** / **Network** / **Ethernet 1 Parameter / Network Config.**

3.5. Managing links

3.5.1. Setting up and releasing links

Generally speaking, once the codec is set up and the transmission interface(s) to be used is(are) configured, it is possible to manage audio links with remote devices.

In the case of a leased line, the link sets up as soon as the physical connection is made, and all parameters are suitably configured: this is a permanent link.

In other cases, transmission links/sessions have to be set, which can be done in two ways:

- "Outgoing call" launched towards a remote device: the procedure is to select a transmission interface, a coding configuration, "dial" the destination to call and then set the call to the remote unit. The detailed operating mode is described further, for the various transmission media, in 4.5 (Setting up a link).
- "Incoming call" received from a remote device: on receiving a call on one of the connected and active interfaces, the codec switches to this interface and processes the call.

In a similar way, ending/releasing a link is either initiated by the remote unit (remote release), or by the operator of the SCOOP 5 (local release).

3.5.2. Auto-redial feature

In LL mode, the unit is always supposed to be linked, and it will transmit and look for a reception signal as long as it is on. In contrast, the IP mode and the ISDN modes are "dial up modes", where a link can be set up and released at will. When it is necessary to hold the link on permanently, outgoing calls may be backed up by using the auto-redial function (for each codec in the double ISDN mode). When it is active and the codec is the initiator of the link, the codec automatically tries to re-set the link in case of an initial failure, or if an established link is dropped for another reason than a local release (i.e. hanging up by the user). The redial capability applies in the following situations:

- If the initial call fails for any reason (e.g. called party is busy); the codec then redials and retries to establish the link.
- The codec can also redial if the link is already established and the link is lost, for any reason else than "local release" (e.g. the remote unit mistakenly dropped the line).
- After a power failure, after rebooting the codec will automatically redial and set-up the link back.

ⓘ *Note that, while "auto redial" is active, an established link can be definitively stopped only by releasing the line on the calling codec side. Otherwise, every time the called party will hang up, the calling codec will redial and reset the link.*

It is possible to program the time period that the unit will wait before redialling after a failed trial, and it is also possible to program the maximum number of times the codec will redial before giving up.

The activation of this function and the configuration of its parameters can be found in the "Auto Redial" sub-menu (**Tools** / **Misc** / **Auto Redial**). In dual codec mode, the function can be activated separately for each codec.

3.5.3. Loop control

In normal operation, outgoing calls are sent or released using the menus and/or the remote control interface. When the loop control function is selected, outgoing calls are controlled by activating or not optically isolated input loops. One loop is available for each codec when in double codec mode. When the input loop is activated (i.e. current is flowing), the corresponding codec establishes a link by calling the last number (or IP address, or SIP URI in the IP mode) previously dialled by the unit. When the loop is de-activated, the codec releases the line and stays idle as long as the loop is not active (except if receiving an incoming call).

ⓘ *In normal operation, it is nevertheless possible to release a running connection by briefly activating ("pulse") the control loop.*

The "auto-redial" feature is implicitly active when loop control is active: the codec tries to keep the link, and automatically recalls the remote unit if the line drops, as long as the input loop is active. The "time before redial" parameter described in the above is also applicable to the loop control mode. On the other hand, the "redial attempts" parameter is not applicable here, because the unit will always try to recover the link, until the loop is left inactive.

ⓘ *Note that, as an important consequence, when using loop control, the termination of a link must always be done on the calling party side by de-activating the input loop. Whenever the line is released by the receiving party, the calling unit will redial and re-establish the link.*

When loop control is active, the input loops are the only means of setting up an outgoing call; setting a call from the menu is not allowed. Hanging up with the keypad is rejected.

ⓘ *Remind that the first step is to set-up the link once in normal mode, and later activate the loop control mode; afterwards the input loop is used to trigger a redial to the previous number.*

In LL mode the link is always active. Loop control, in this case, can be used to control a call via a backup IP or ISDN link: *see further in 3.5.4, Setting up a backup link*.

3.5.4. Setting up a backup link

The SCOOP 5 has the capability for backing up a permanent leased line audio link thanks to a switched link (ISDN, IP, POTS...).

A typical example of this application is shown in the following diagram, where a permanent leased line is used for transmitting a radio programme from a studio to a transmitter. A codec is installed on each side of the link, and each codec is also connected to a "backup" link (ISDN, IP, etc.). In this way, transmission via this line can be used as a temporary back-up in case the normal permanent link fails. Once the problem on the normal line is solved, the backup connection can be released and normal operation in LL mode is restored.

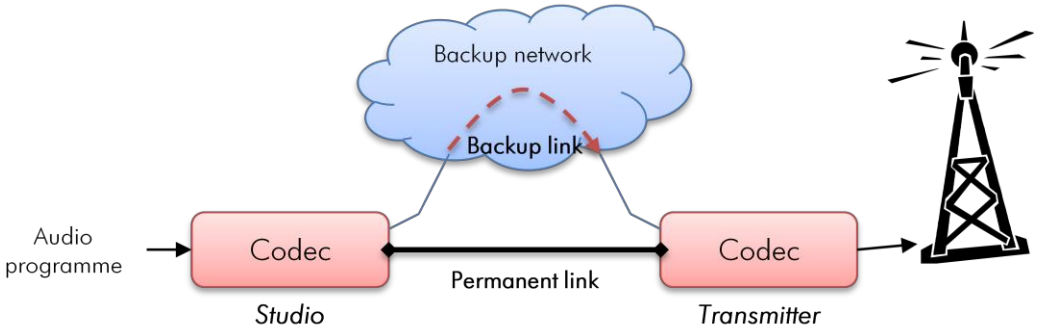


Figure 6 – Backup link

The SCOOP 5 makes such back-up easy because both transmission interfaces (normal leased line and backup interface) are available in the same unit. Moreover, the SCOOP 5 helps to automate the switchover process, especially on the transmitter side, where most often quick human intervention is not possible. The following describes one way of configuring such a system, and details the resulting backup process.

Basic principles

When a failure of the leased line is detected, one of the two codecs has to switch to "backup" mode, and then call the other unit over the backup network access. On receiving this call, the remote codec will have to switch as well to the backup network interface and accept the call. The audio transmission in then provided over the backup link. Later on, when the leased line comes back to normal, the backup connection can be released and both codecs have to switch back to LL mode and normal operation.

We designate as "caller" the codec which switches first and initiates the call over the backup network, and "receiver" the codec which switches on receiving such a backup call.

Caller codec: setup and operation

For the switch to backup (following a failure on the leased line), two methods can be used:

- Either the switchover is "manual": an operator on the studio side can switch the studio codec to backup mode and launch a call to the other site via the backup medium. For this method, no special preliminary setting is needed.
- The other method is to use the "loop control" feature as described above in section 3.5.3. In this case, the whole switchover sequence can be executed by simply activating the input loop: mode switch for the backup network access, backup call. Conversely, releasing the input loop brings the unit back to normal operation in LL mode.

For this latter method, the best situation is when the leased line status is monitored by an alarm contact that closes when the line is down. This alarm can be used for activating the control loop, so that the studio codec quickly and automatically switches over to backup mode (and also comes back to normal when the leased line recovers); no manual intervention is needed in such case.

In order to use this operating mode, two settings have to be done on the "caller" codec:

- Select the network to use for backup: menu **Tools** / **Misc** / **LL Backup** / **Backup Call**
- Activate the loop controlled backup: menu **Tools** / **Misc** / **Loop Control** / **Backup**

ⓘ *Important notice: when switching to backup mode the number called is the last dialled! For this reason it is important to dial and call the backup destination in the desired mode at least once, before setting the unit in its normal LL mode.*

Receiver codec: setup and operation

The setup is easy on the "receive" side of the backup link, as it is somewhat "passive". The "backup receive" mode must be activated, via the menu **Tools** / **Misc** / **LL Backup** / **Backup Receive**.

Once this is done, when the codec is operating in LL mode, on receiving a call on the backup interface, the codec switches to the corresponding network mode, answers the call and the connection is established with the remote codec. Later, when the temporary connection is released (by the calling party), the "receiver" codec automatically switches back to LL mode, and normal operation is resumed.

ⓘ *Reminder: in LL mode the codec accepts no incoming call whatever the network interface, unless the backup receive mode is active.*

ⓘ *Conversely, once this is done, the codec accepts incoming calls regardless of the network interface.*

ⓘ *Thanks to the automatic coding algorithm detection (5AS in ISDN mode or SIP/SDP in IP mode), the coding setup is not critical on this "receiver" side. However, for increased safety in ISDN mode one may prefer to force the configuration (by de-activating the 5AS).*

Notes regarding the automatic switchover

Note that, when switching from LL to backup mode, a separate set of parameters is recalled. As a consequence, the coding configuration can be made, if desired, totally different in the back-up mode.

Of course, both units should be configured in an adequate configuration for each transmission mode, and then each codec can be set in the LL mode to start the normal operation.

ⓘ *For the backup configuration, **do not set the units in double codec mode**. Otherwise, the automatic switchover will not work properly.*

3.6. First level maintenance

3.6.1. Internal configuration

All the configuration is done in the factory, and/or it can be changed by means of the keyboard/display interface, without having to open the unit.

However, a few settings can be done internally by setting jumpers:

- It is possible to prevent one alarm type to light on the red alarm LED on the back of the equipment.
- It is possible to disable the standby mode (in which case the device is always in operation as long as the mains power is present).

Please consult us for such operation! We remind that unduly opening the unit can void the warranty. **In any case, opening the unit may expose live parts and is hazardous. Never open or maintain the internal parts without first disconnecting the AC supply.**

3.6.2. Analysis of malfunctions

The following table indicates the detected alarm conditions and their classification:

Alarm condition	Internal	External	Minor[1]
Power or fuse fault	X		
Bad start-up of a microprocessor, or interface fault detected on start-up	X		
Overload on an audio input			X
Fault on AES/EBU audio input		X	
Decoder synchronisation error		X	
Network clock fault[2]		X	

Table 7 - List and classification of alarm conditions

Excluding the case when an internal failure disables the management micro-controller, messages are displayed to indicate the anomaly, or the fault can be searched using the menu (**Tools** / **Status**).

[1] Minor alarms are readable on the display, but do not trigger alarms (contacts and LEDs)

[2] Fault of the network clock source currently used for synchronisation (X21/X24 main port or secondary port)

3.6.3. Using the test loops

The test loops accessible from the "**Tools** / **Maintenance** / **Tests** " menu can help improve the analysis of a problem:

- In order to check if the audio part functions correctly, use the "Audio" loop and check if the audio is OK at the output.
- To check if the coding part functions correctly, activate "Loop 3" [1] and check if the alarm disappears (and the decoding indicators come back to normal), and if the audio is present at the output.
- "Loop 2"[2] sends back to the remote codec the compressed data received from the network (see the description of test loops in 2.5.8, Test functions). In this way, it is possible to test the integrity of the transmitted data and/or check that the remote codec works properly.
- The decoder out to encoder in loop ("Audio feedback" loop) can be used for overall functional check, and also for aligning the overall chain.

Note: not all test loops are available for all the network interfaces.

3.6.4. External alarms

In leased line mode, a clock fault is one typical cause of an external alarm. This can be due to:

- complete loss of the X24/V11 interface, due to a failure of the transmission line;
- a failure of the transmission device connected to the codec;
- an incorrect clock frequency (i.e. incompatible with the codec configuration).

On the other hand, in case of a decoder alarm with no clock error, possible causes are:

- lack of signal received from the X24/V11 interface, due to a failure of the transmission device connected to the codec, or a transmission failure in the network;
- a fault in the remote codec, or else the remote codec has an incompatible configuration;
- transmission errors causing erratic alarms.

Errors such as "AES error" and "AES sync loss" can frequently be seen, even when the unit is configured to use the analog inputs. This is because the AES output is always active, and by default "genlocked" to the AES input. To avoid such undesired alarms:

ⓘ *When not using digital audio interfaces, set the digital audio sync in "Master" mode (**Setup** / **Audio** / **Digital** / **AES Synchro** / **Master**)*

[1] Note: this loop is not available for all transmission configurations

[2] This loop is not available for all transmission configurations

3.6.5. Backup reset

This procedure should be applied to recover control over the unit if it is in a blocked status, in which it is impossible to access either the front panel interface or the html page.

After this reset, the SCOOP 5 will be brought back to its "factory" configuration with all parameters in their default value, especially: blank password, Ethernet interface set for auto link mode, DHCP client enabled.

Go through the following steps:

- Get the *reset_scoop.bin* file (to be downloaded from our www.aeta-audio.com web site, a link can be found on the page dedicated to SCOOP 5).
- Copy this file *without changing its name* on a USB memory stick.
- Plug this stick into the USB socket on the rear panel of the unit (while it is off).
- Power on the SCOOP 5.
- After the initialization phase, SCOOP 5 will go back to its "factory" settings.
- Remove the USB stick (no matter with or without power on), before restarting the SCOOP 5, otherwise all your possible new settings will be deleted again.

ⓘ *Note: existing profiles are not deleted by this procedure.*

4. Detailed operating mode – User interface

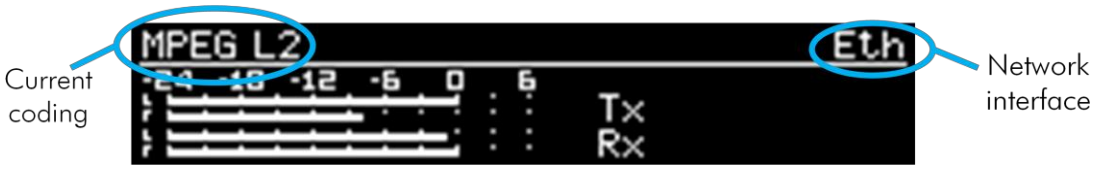
In local mode, the unit is operated thanks to a keypad and display on the front panel. The display is a backlit graphic LCD.

Operating from the keyboard can be protected by a password (numeric sequence, that can be programmed or erased by the user. In such case, the operation is in a "restricted" mode, without direct access to settings or dialling. Only the profiles menu is accessible, for loading a profile or call a remote profile (*see later in 4.7,* Restricted operation mode *for more about the restricted mode*).

4.1. Equipment start-up

When the unit is powered but in standby (blue LED on), press the ⏻ key for at least 3 seconds to start up the unit.

During start-up, the unit displays temporary messages. This initialisation lasts around 30 seconds. Then the main menu is displayed.



On this "base screen", transmission and reception audio levels are permanently monitored for the two audio channels.

At this time, the unit can be operated and receive calls. It is also possible, using the menus, to change settings and/or set up calls.

ⓘ *If the unit has been set in "restricted access" mode, for recovering full access to all menus type directly the password (numeric sequence), and the device unlocks as soon as the last figure of the password is pressed.*

See further in 4.7 how to use and manage this restricted access.

4.2. Principles for the navigation

From the base screen, if pressing the Esc key the display switches to the highest level menu, called "root menu"[1].

Pressing Esc again brings back the base screen.

The LED indicators and the main keys surrounding the LCD are described above in 3.2.1, Front panel

The unit proposes a tree structure of menus, and the arrows around the OK key are used to navigate through the menus. The OK key is used to confirm choices or enter values for parameter, while the Esc key allows to come back up to the upper menu level. Repeatedly pressing this key brings you up to the root menu, and the base screen.

From the root menu, you can enter one of the three main menus by using the arrows and pressing OK to enter the highlighted menu.

From these main menus, the screen is organized in the following way:

The sub-menus have the following aspect:

At the end of a "branch", lists of choices appear in this way:

In all cases, the principles are as follows:

- Use the arrows to browse sub-menus or choices.
- Enter the highlighted sub-menu by pressing OK or the ▶ key.
- Move to upper level with the Esc key.
- Validate a choice or entered value using OK
- Give up a selection or action by pressing Esc (repeatedly as needed)

ⓘ *After some time without pressing any key (about 30 seconds), the display comes back to the base screen.*

[1] You can also press the OK key

4.3. Dialing and text input keypad

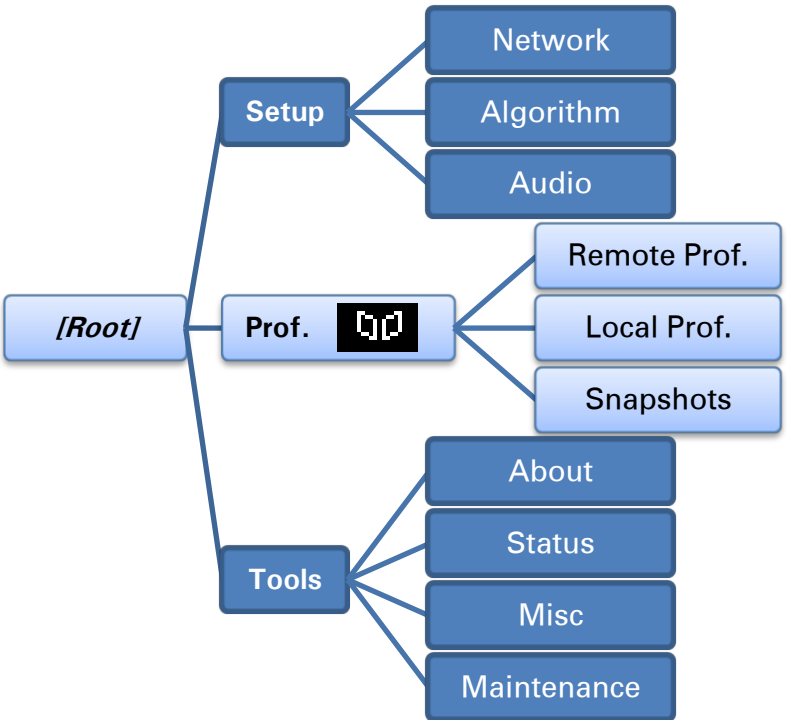
The keypad is used for entering numbers and/or texts. It is used in a similar way to a mobile phone:

- The keypad works in "numeric" only mode, or in "alphanumeric" mode (where both numbers and letters can be entered).
- In numeric mode, only the numbers are used, and the "*****" key (as a separation between ISDN address and sub-address, or as a dot in a numeric IP address).
- The "alphanumeric" mode allows text entry. A letter can be entered by pressing repeatedly a key; for example, a "B" is entered by hitting the "**2**" key two times (sequence A, B).
- Use the **#⇑** key to switch between "numeric", "alphanumeric (lower case)" and "alphanumeric (capital)" modes.
- The "**0**" and "**1**" keys provide access to various special characters which are not all printed on the keypad: "**@**", "**:**", "**-**", "**_**"…

4.4. Description of the menus

The root menu shows the following three main menus:

- **Setup** configuration of the codec
- (profiles): access to configuration memories or "profiles"
- **Tools** Maintenance and miscellaneous functions, access to status information



The « **Setup** » menu leads to the following sub-menus:

- **Network** selection and configuration of the network interface and parameters
- **Algorithm** selection and configuration of the coding algorithm
- **Audio** configuration of the audio interface and parameters

The menu for profiles leads to the following sub-menus:

- **Remote Prof.** selection and edition of remote profiles
- **Local Prof.** selection and edition of local profiles
- **Snapshots** selection and edition of snapshots (memories for audio parameters)

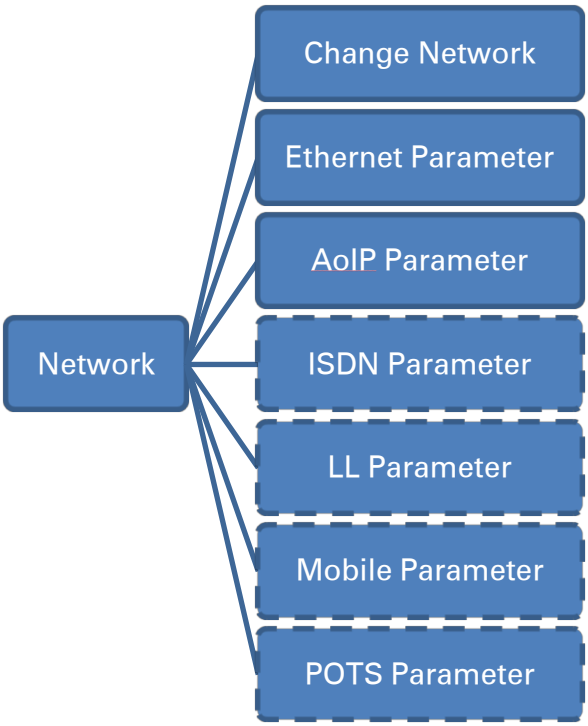
ⓘ *This menu is the only branch of the menu tree that is accessible even in restricted mode.*

The « **Tools** » menu is itself divided in sub-menus:

- **About** display information on addressing and firmware version
- **Status** display information on the status or operation errors
- **Misc** miscellaneous functions
- **Maintenance** maintenance and test functions

The following diagrams show the various sub-menus and the available parameters.

4.4.1. Network sub-menu



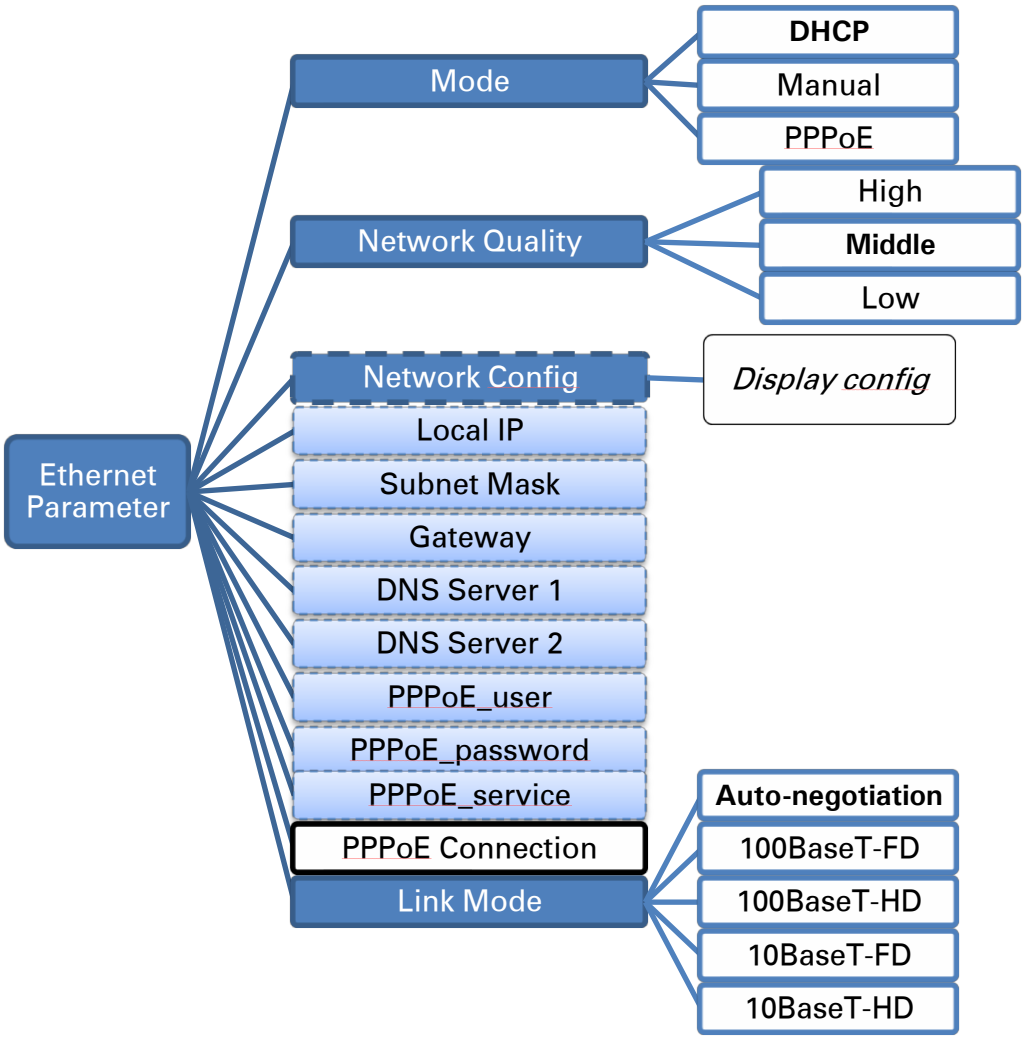
The **Change Network** sub-menu shows a selection among the available network interfaces. This is how you select the "default" interface, which is used for an outgoing call. This menu must be used before setting a call (except if the default interface is already the desired one). The current interface is recalled on the right side of the screen:

The other sub-menus provide access to the settings for each existing interface. This is why some interfaces (*with a dashed frame outline on the picture above*) may not be visible in the sub-menu depending on the unit version, if it does not include such interfaces.

ⓘ *If you have plugged a 3G or LTE "mobile key" on the USB interface, you can also see this module in the list, and you can select it just like another network interface, designated as "Mobile 1 (Ext.)". If more than one mobile device is available (example: presence of an integrated module and also an "external" device connected on the USB socket), then each is identified in a unique way.*

ⓘ *In a similar way, if an Ethernet adapter is plugged on the USB Interface, you will see in the list a new interface, named "Ethernet 1". The corresponding sub-menu has the same structure and settings as the sub-menu "**Ethernet Parameter**".*

Ethernet Parameters



Notes:

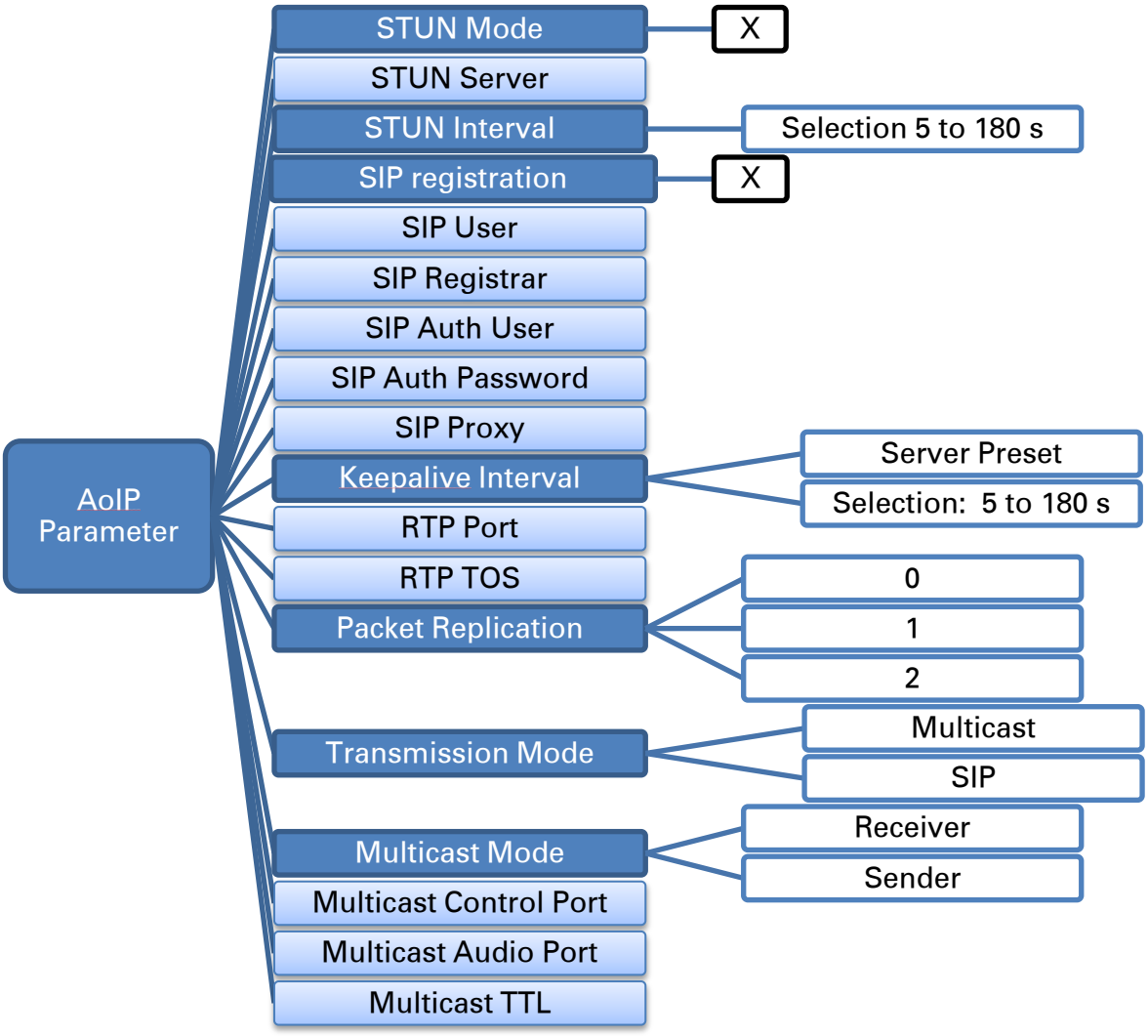
- **Mode**: the current setting is recalled on the right side. The selection is operative immediately after selecting an addressing mode.
- **Network Quality**: select here the expected quality of the transmission via the Ethernet interface. The unit switches to internal settings suitable for this quality level.
  For instance, if you select "**Low**", the codec sets a large size for the reception buffer, in order to stand a higher jitter. As a counterpart, the latency will be higher as well.
  Conversely, if you select "**High**", the buffer is smaller and the latency is lower, but the system is more vulnerable to possible jitter (fluctuations of the network transmission latency).
- **Network Config**: only appears when DHCP is active, in which case this sub-menu displays the addressing parameters that the SCOOP 5 got from the server:

  Using the arrows, you can browse the various settings: local IP, sub-network mask, etc. *Exit this display with the* Esc *key.*
- If DHCP is not used, several menus allow you to enter the needed settings, such as **Local IP**, **Gateway**, etc. (*lighter background on the picture above*)

- **Link Mode** is related to the physical Ethernet interface; the default setting "**Auto-negotiation**" is suitable for almost all situations.

AoIP Parameter

This sub-menu groups the settings related to the Audio over IP (AoIP) transmission. It is relevant for the Ethernet interface but also for using a mobile access network, be it with the integrated module or an external module via USB.



Notes:

- **STUN Mode**: enable or disable STUN. *In this way you can disable STUN without erasing the STUN server address.*
- **STUN Server**: enter the address of the STUN server, either in numeric form or with a symbolic name. This kind of server allows SCOOP 5 to discover its public IP address when it accesses the Internet through a router with NAT. *You can use AETA's server:* ***stun.aeta-audio.com.***
- **STUN Interval**: interval between two queries to the server; usually you can keep the default setting of 15 s.

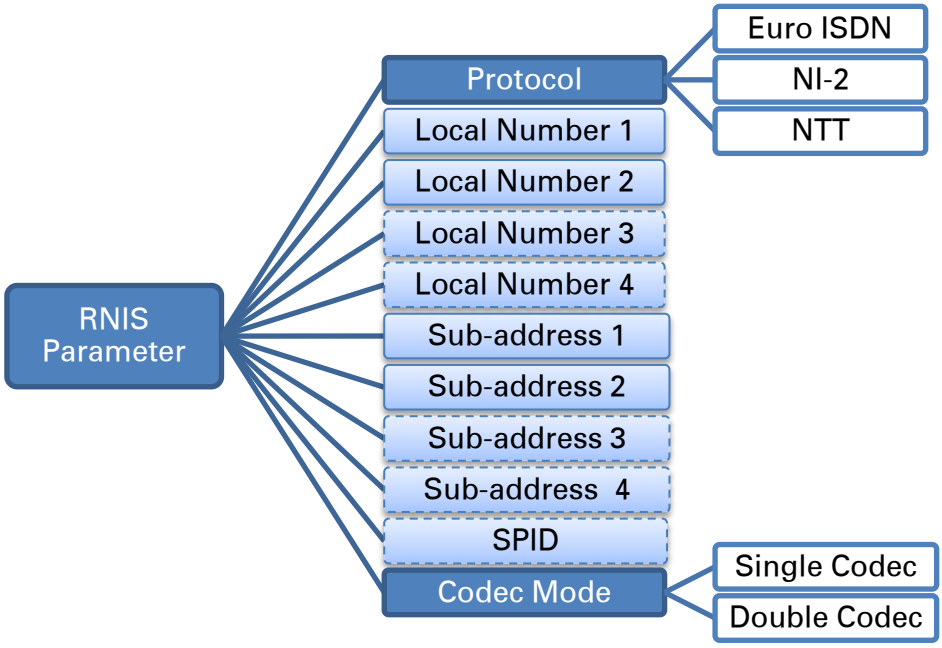
- **SIP Registration**: enable or disable the registration on a SIP server. The following menus SIP xxx are used to enter the parameters of the account on this server, when using such service. These parameters should be provided by the administrator of the SIP server. The **SIP User** field usually corresponds to a number that a remote party can call to initiate a link with the codec when it is registered on the server.
  With this **SIP Registration** sub-menu, *you can disable the registration and hence stop using a SIP server, without having to erase account parameters; they stay ready for later use.*
- **Keepalive Interval**: sets the interval between two re-registrations on the SIP server. This setting can be used to make the registration renewal more frequent, but the server always imposes a maximal period; for this reason the default setting "**Server Preset**" is recommended except for specific cases.
- **SIP port**: the default port for SIP signalling is *5060* (UDP), you can enter a different value here.
  *Note: this port number is valid for the SCOOP 5 itself, not the server. For the server side, if a non-standard port is used, e.g. 5070, the "**SIP Registrar**" field must specify the port after a colon like in "mySIPserver:5070".*
- **RTP Port**: the default port for RTP/SIP audio streams is 5004 (UDP), but you can enter a custom value here if desired.
- **RTP TOS**: if the transmission network supports DiffServ to manage the media priority, you can set here the type of service which is assigned to the stream transmitted by the codec.
- **Packet Replication**: (see also 2.1.1 and page 8 regarding this feature) you can select here the setting for packet replication:
  0 => standard mode without replication (default setting)
  1 => packet replication without interleaving
  2 => packet replication with interleaving: the second packet is delayed; robustness is improved but latency is increased.
- **Transmission Mode**: select here between SIP (unicast) or using multicast (see also above in 2.1.1, IP multicast). *The following sub-menus exclusively apply to the multicast mode. Besides, the SIP settings are not relevant for this multicast transmission mode.*
- **Multicast Mode**: if the selected mode is Multicast, the transmission is unidirectional. Select with this menu if the codec must be *sender* of a multicast stream, or rather *receiver* of such stream.
- **Multicast Control Port**: this port number (UDP) is used for sending control packets accompanying the multicast stream. Default value is 6000.
- **Multicast Audio Port**: this port number (UDP) is used for transporting the multicast audio stream. Default value is 6001.
  *Note: this setting is not related to the **RTP Port** setting valid for RTP/SIP.*
- **Multicast TTL**: TTL value applicable to the RTP multicast audio stream. Default: 254.

LL Parameter

Notes: **Interface**: select the X24/V11 used for transmission; interface 1 is the default interface.

ISDN Parameter

This sub-menu is not available on versions with no active ISDN interface.

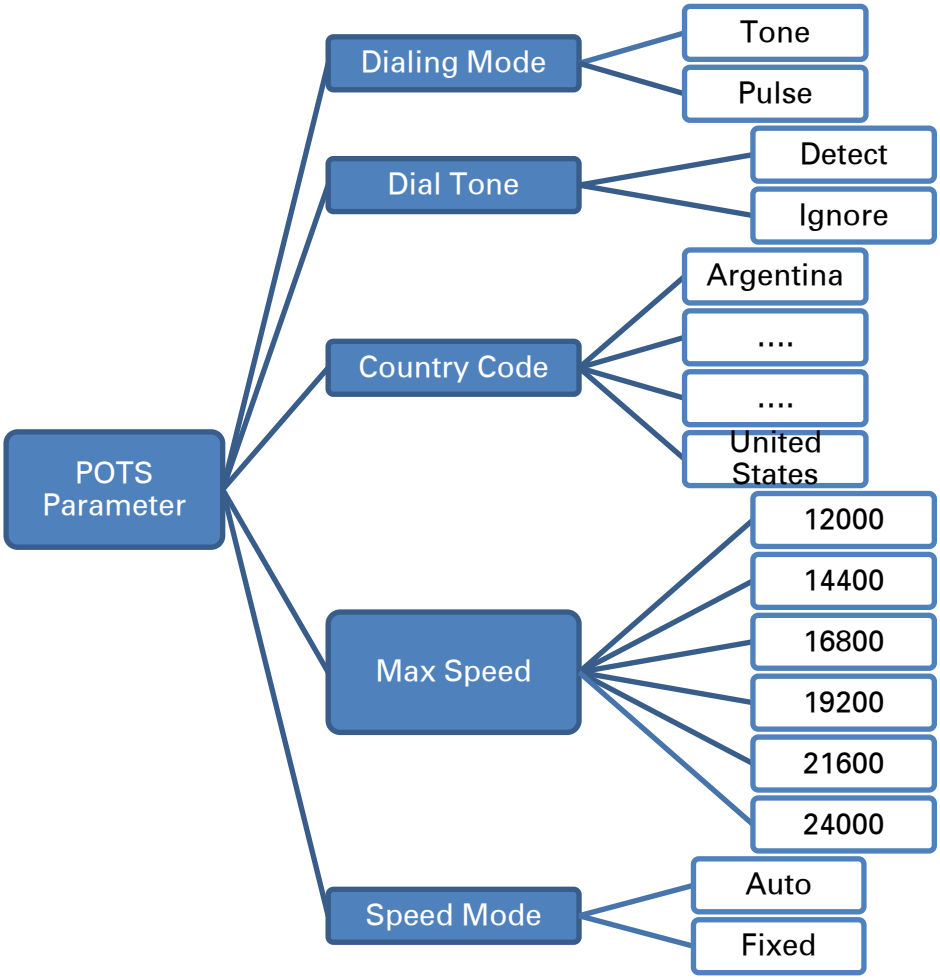


Notes:

- **Protocol**: NI-2 is used in North America, NTT is the protocol of the Japanese operator NTT, and Euro ISDN (or ETSI) can be used in the majority of the other countries (*see more details on page 9, Network protocol*).
- **Local Number** and **Sub-address**: these settings are irrelevant for the NI-2 protocol. *Their meaning and use is detailed further, page 66, in 4.5.1,* Setting up a link in ISDN mode.
- **SPID**: this setting only applies to the NI-2 protocol, and is not presented for the other protocole. See page 66, in *4.5.1,* Setting up a link in ISDN mode, more about this setting.
- **Codec Mode**: selection of either the normal "single codec" mode or the "Double codec" mode. In this latter case, it may be important to set a distinctive local number + sub-address combination for each of the two first channels. This is detailed *page 68, in 4.5.2,* Setting up links in double ISDN codec.

POTS Parameter

This sub-menu is only visible on units with the POTS option.



Notes:

- **Dialing Mode**: tones are used by default (DTMF), but for specific cases (old PBXs, etc.) it is possible to select pulse dialing.
- **Dial Tone**: choose "**Detect**" so that the codec waits for the dial tone before dialing the numbers. The dial tone may be specific with some particular equipment or networks, and may not be detected by the SCOOP 5; in such case, disable this function ("**Ignore**", and the codec will dial the numbers just after picking up the line. *This latter choice also works correctly with most modern PBXs.*
- **Country Code**: this choice will set the internal modem for the characteristics applicable in the selected country. The full list is not shown above as it is quite large! For a country not found in the list, usually it is possible to select a country with similar standards (such information may be found from local specialists).
- **Max Speed**: in nominal use, the POTS codec runs at 24000 bit/s, but it can automatically adapt according to the line quality, and change the bit rate if required at the beginning and during the link. With this setting, you can constrain the bit rate to a lower value. Reason for this: on a line featuring a variable quality[1] you can avoid undesirable changes during the communication by "limiting" the modem to a speed that you know to be stable. The speed may nonetheless go down further if the quality decreases too much.

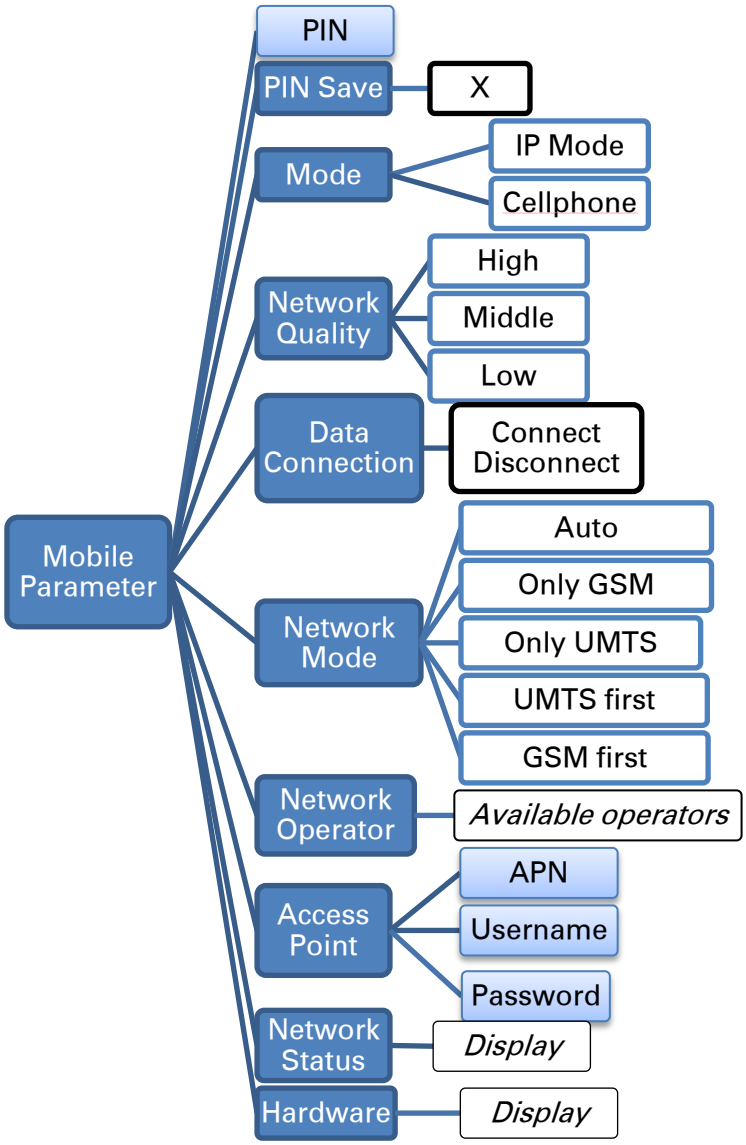
[1] This can happen for example when the line suffers a strong crosstalk from neighboring lines.

- **Speed Mode**: for a line/transmission with variable quality, and/or to avoid repeated changes of the bit rate, with this setting you can force a fixed bit rate. In such case, the "Max Speed" above setting becomes a fixed value. Be aware you must be sure that transmission conditions will not require a lower speed; otherwise you incur a synchronization loss in such eventuality.

Mobile Parameter

This sub-menu is only visible on units with the "wireless" option.

If more than one mobile access module is available (example: integrated module and "mobile key" on the USB socket), then one such sub-menu is presented for each module.



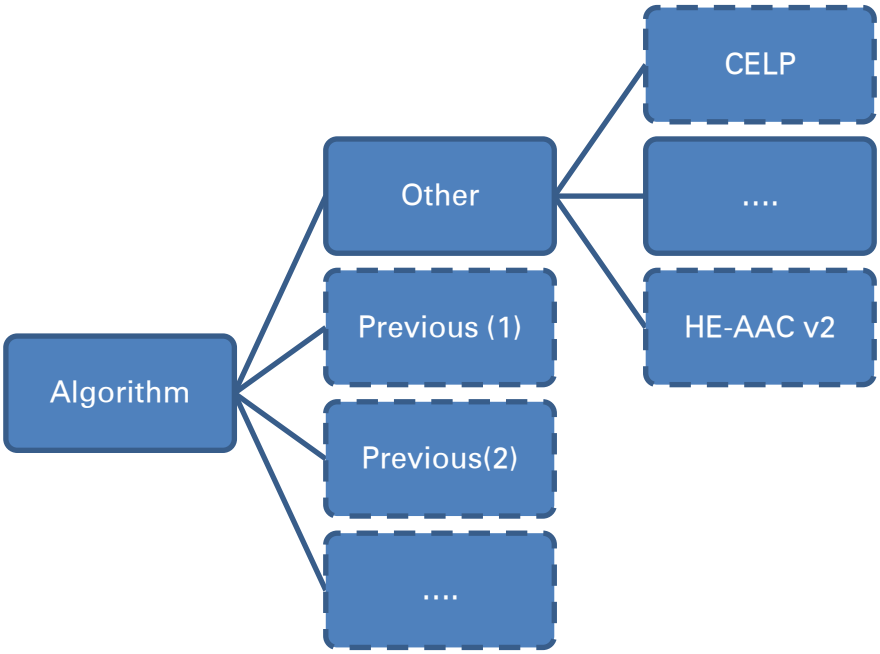
Notes:

- **PIN**: enter the PIN code allowing to use the SIM Card and access the network. Entering this code is the first step necessary to operate the mobile access, except if the code for the card is empty or disabled.
  The usual safety rules apply to this setting, so be careful not to lock the SIM card with repeated mistakes!
- **PIN Save**: if this option is checked, the unit saves the PIN code and unlocks the SIM card automatically when the unit is switched on.
  It is recommended to disable this feature before changing for another SIM card (and possibly another PIN). However, the unit will not make more than one trial if the memorized code is wrong.

- **Mode**: select the desired transmission mode: "**IP Mode**" for transmission of data packets (IP protocol), also called "PS" mode (for Packet Switched), or "**Cellphone**" for the telephone mode, which can also be wide band "HD Voice" when all conditions are met (*see above in 2.1.5, Mobile voice mode – HD Voice*).
  Note: the "cellphone" mode is not available for a "mobile key" device on the USB interface, because such devices only support IP data transmission.
- **Network Quality**: select here the expected quality of the transmission via the mobile interface in IP mode. This is the same setting as for the Ethernet interface (see above in 50, Ethernet Parameters).
- **Data Connection**: applies to the IP mode; use this to activate the data connection, or disconnect it if it is active. This is similar in function to connecting an Ethernet cable on a wired network: by activating the connection the unit is linked to the Internet and it can send and receive IP data. For instance it can receive an incoming call. It can also initiate an outgoing call.
- **Network Mode**: select the network technology. "**Auto**" is the normal choice, which provides operation on 3G/3G+ as long as the conditions allow it, but fallback to GSM if 3G is not available or the radio link is too poor. It is also possible to force either the 3G/3G+ modes ("**Only UTMS**" selection), or the GSM/2G mode ("**Only GSM**" choice).
- **Network Operator**: normally the unit automatically selects a suitable network. If you want to force a specific operator (which can happen when using the unit abroad), this menu presents the list of available network so you can manually select one. *But not all networks are actually allowed, this depends on the policy of the SIM card's provider!*
- **Access Point**: this sub-menu allows to enter the necessary parameters for data connections (IP mode): APN, user name and password. The essential setting is the **APN** (Access Point Name), mandatory for accessing the mobile IP network. SCOOP 5 automatically sets the APN for many operators, but you may have to enter this yourself, or change the default setting. The other parameters can usually stay empty; otherwise they must be provided by the network operator, along with the SIM card and the APN.
- **Network Status**: information on the mobile network (for the current connection): network/operator identification, technology (GSM, EDGE, UMTS, HSDPA, HSPA, etc.), radio signal strength. *Note: bars on the base screen also show the mobile reception level)*
- **Hardware**: information on the internal mobile access module (type, firmware, IMEI) and about the SIM card (IMSI, subscriber number).

4.4.2. Algorithm sub-menu

This sub-menu allows to select the desired configuration for an outgoing call, using the current transmission interface.

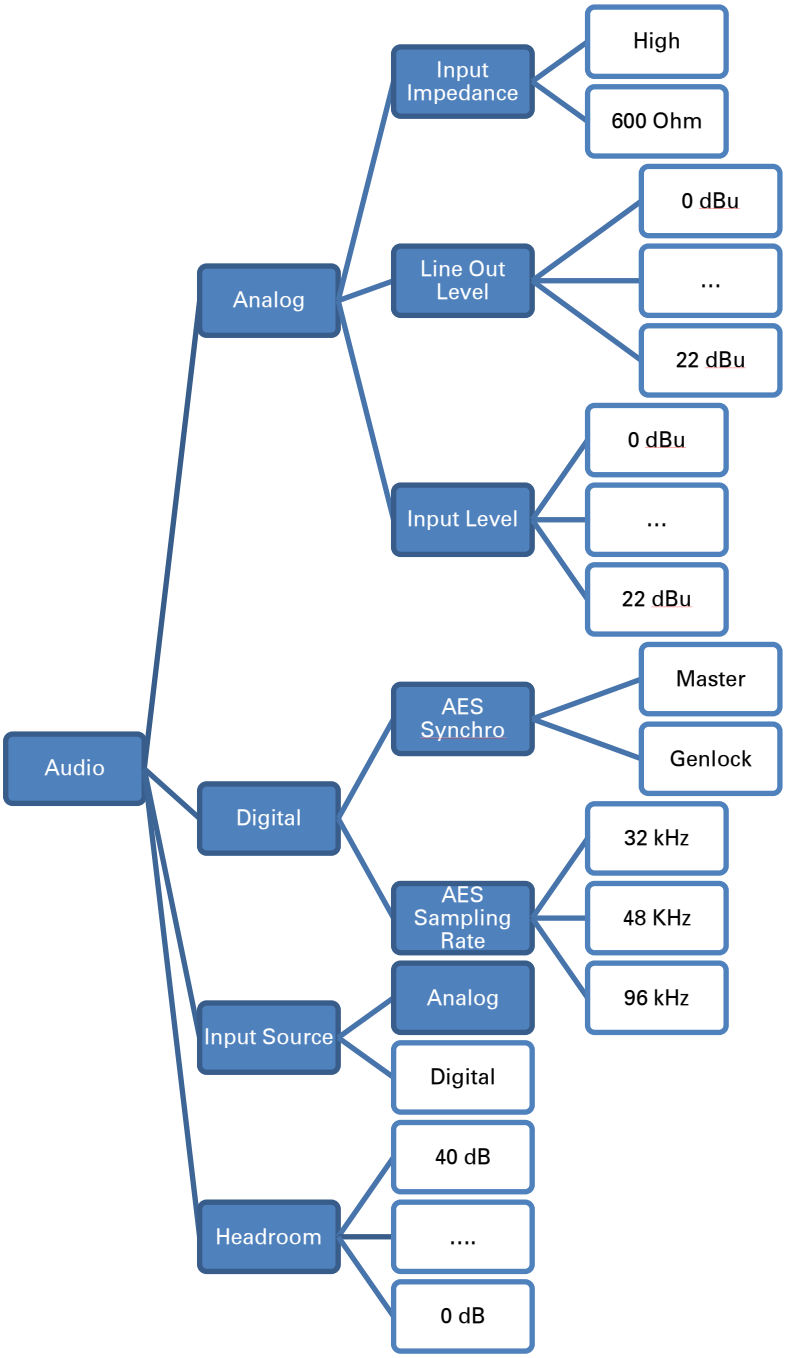


It is possible to quickly recall the last used configurations ("Previous" above), scrolling the list and selecting one of these recent settings.

Otherwise, selecting "**Other**" you get the list of the coding algorithms available for the current network interface and the installed options. Choices that are not compatible with the network interface or not installed are not displayed.

After selecting the coding algorithm, for some of these you must select additional parameters, such as: channel mode (for stereo-capable coding), sampling frequency and bit rate (for MPEG L2 and AAC coding), protection mode (only for MPEG J52 and CELP)

4.4.3. Audio sub-menu



Notes:

- **Analog**: settings for the balanced analog inputs/outputs
- **Input Impedance:** configuration of the analog inputs, by default high impedance. You can switch to a 600 Ohm input impedance instead.
- **Input Level** and **Line Out Level:** gain setting for (respectively) the inputs and the outputs. The parameter is the maximum level (absolute level expressed in dBu) of the interface. For an input this is the clipping level, which when applied at the input produces a transmitted signal at 0 dBFS. For an output this is the maximum output level, produced when receiving a signal at maximum level, i.e. 0 dBFS.

- **AES Synchro**: the default choice is "**Genlock**": digital interfaces (input and output) locked onto the digital audio input. This choice is suitable for most cases, when the codec should slave to the system clock. However, it is possible, if required, to rather synchronize these input/output from the internal clock of the codec: so-called "**Master**" mode (*see also in 3.3.3 about the installation*).
- **AES Sampling Rate**: sampling rate to be used in the "Master" mode. In the Genlock mode, this setting has no direct influence because the codec "follows" whatever sampling frequency is set by the AES input.
- **Input Source**: select the audio source to be used by the encoder: either the analog inputs, or the AES input. *Even if you use the analog inputs, the AES input can still be connected in order to synchronize the AES output.*
- **Headroom**: this setting defines the reference level for the program level display. This display is relative to a "0 dBr" reference which can be set here. The "Headroom" parameter is the available headroom for a signal displayed as 0 dBr, i.e. the difference between the reference and the 0 dBFS maximum level. For instance, a headroom of 10 dB means that the reference is set at -10 dBFS.

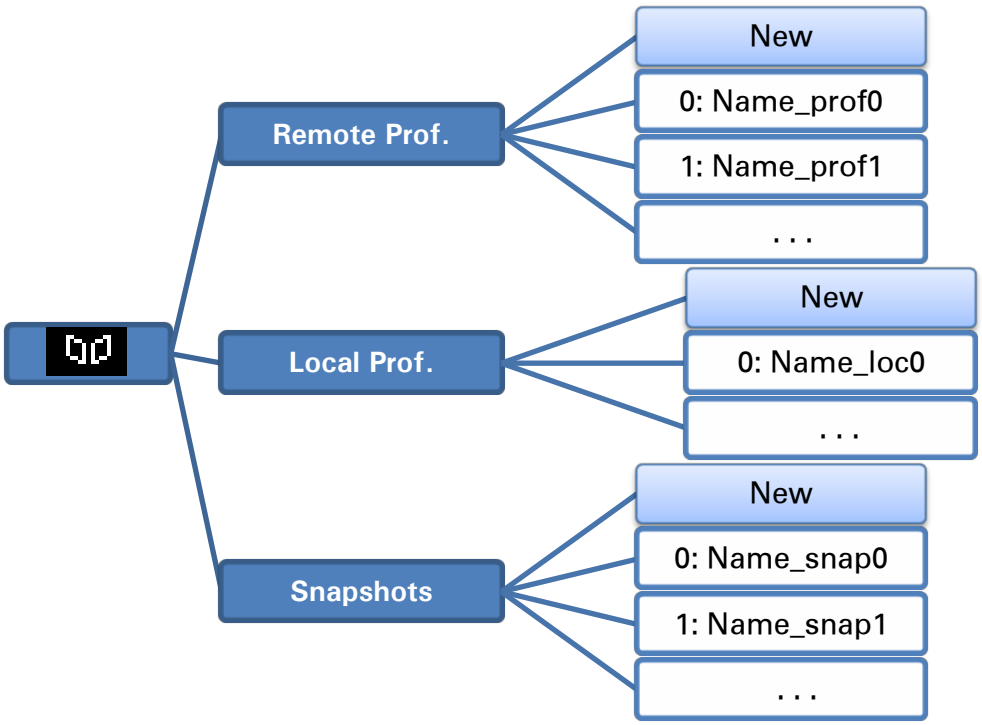
ⓘ *On the display, the "OVL" clipping indicator shows that the absolute maximum level is reached (0 dBFS), regardless of this headroom setting. This is not necessarily the top of the bargraph scale, which is +6 dBr (relative), and whose absolute level depends on the "headroom" setting. Example, if the headroom is set at 10 dB: then the reference is -10 dBFS. The bargraph can go up to +6 dBr, which is -4 dBFS. At this stage the "OVL" indicator does not yet trigger, as it reacts at 0 dBFS.*

Beware of the differences between the three level measurements!

- Absolute digital levels in dBFS for the AES input and output, internally (encoder/decoder) and for transmission.
- Displayed levels in dBr, relative to an adjustable 0 reference. *The relationship between the display and the dBFS level is controlled by the "**Headroom**" setting.*
- Absolute analog levels in dBu for the analog inputs and outputs. *The relationship between the absolute analog and digital levels is controlled by the "**Input Level**" and "**Line Out Level**" settings.*

4.4.4. Profiles menu

This is the only menu available in restricted access mode.



"Profiles" are configuration memories that can be recorded and later recalled at will (*see their description in 2.5.7 above*). For each profile category, a sub-menu provides the list of profiles already recorded in the SCOOP 5 memory, each with its index and name.

On the top of the list, the "**New**" choice allows you to create a new profile:

- Enter a name for this new profile, then OK. For a "local profile" or a "snapshot", the current settings (respectively network settings or audio settings) are recorded into the new profile with the name you just entered.
- For a remote profile, first you can elect to include or not the coding parameters: select "**Current**" (include the current codec settings) or "**None**" (do not include an algorithm with this profile).
- Then you can enter the dial number (or IP address, or SIP URI), then OK. *You can also leave this field empty; in this case the profile is used for memorizing only coding parameters, rather than a remote number.*
- Lastly, for ISDN and a setting exceeding 64 kbit/s, you must enter an additional number (or more).

To use the profiles, just scroll the list, select one of the recorded profiles, and press OK: a choice is presented:

- **Load** this profile, recalling the memorized parameters;
- **Call** the remote destination in the profile (choice available only for a remote profile): in this case the codec recalls the memorized parameters, and then sets the call to the memorized number(s);
- **Delete** the profile (not available in restricted access mode);
- **Update** the profile (for a remote profile): you can edit an existing profile, going through the same steps as for creating a new remote profile. *This choice is not available in restricted access mode.*

4.4.5. Tools / About sub-menu

This sub-menu provides a list of data about the identification and status of the unit. Scroll the list using the arrows and access all the information:

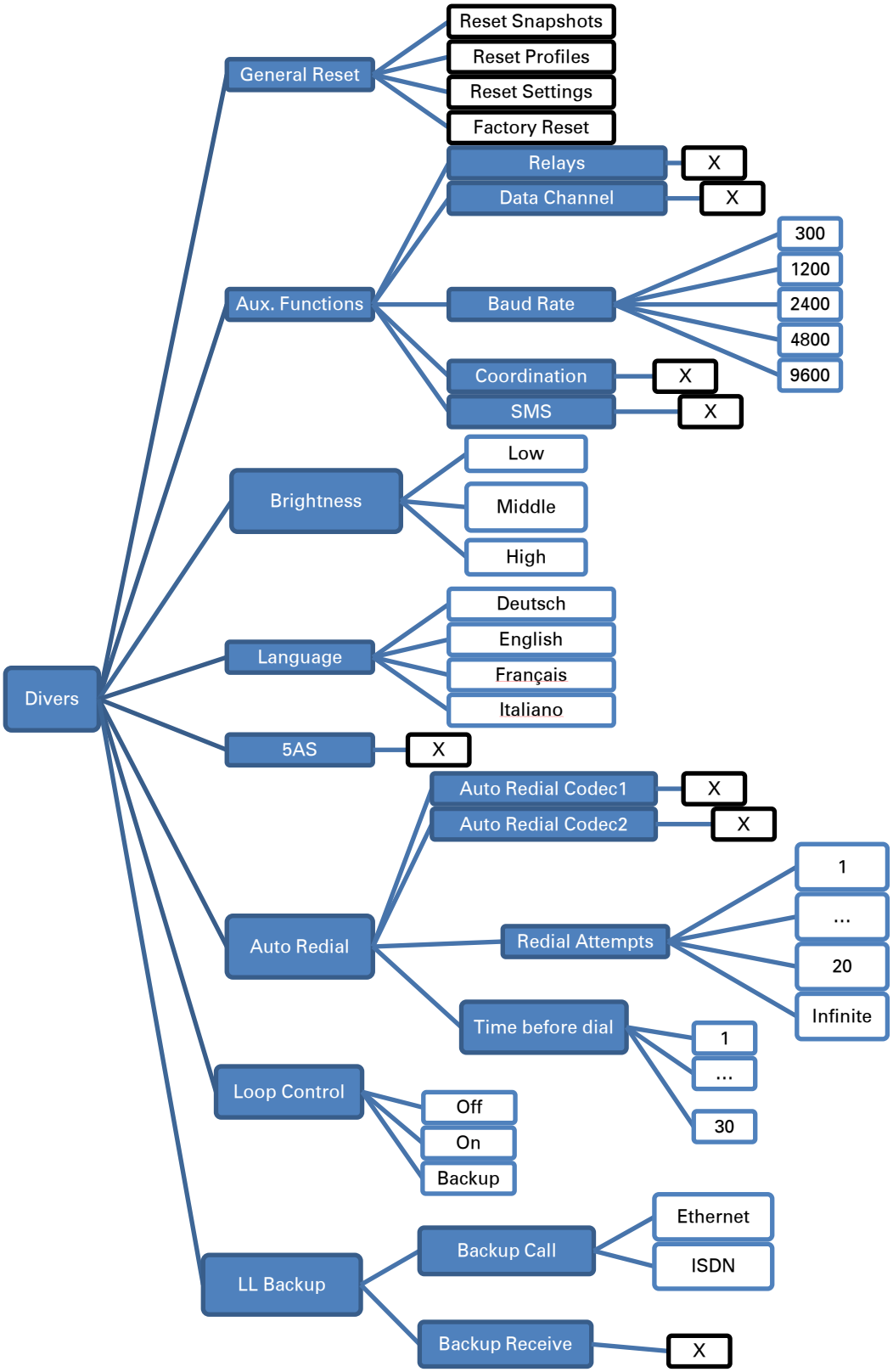
- **Local IP**: IP address of the active IP interface (Ethernet or mobile depending on the current one).
- **Public IP**: public IP address, when SCOOP 5 accesses the Internet through a NAT router. This address is usually detected thanks to a STUN server (configured via the "AoIP parameters" menu, see in 4.4.1).
- **Version**: SCOOP 5 firmware version global identification. *Along with the unit serial number (shown on the label stuck on the rear panel), this is part of the essential information to provide whenever you contact AETA for support or maintenance issues.*
- **EIM Version**: version number for the audio via IP module.
- **SW Build**: date of firmware generation.
- **MAC Address**: of the main (integrated) Ethernet interface.
- **Ethernet Setup**: current setting of the main Ethernet interface (10 or 100 Mbit/s, half/full duplex…)
- **SIP-Registrar**: address or domain name of the SIP server, if one is configured
- **SIP Status**: shows whether the codec is registered on the SIP server.

4.4.6. Status sub-menu

This sub-menu provides the list of alarm conditions detected by SCOOP 5. If there are more than two (in which case vertical arrows are shown on the display), scroll through the list using the vertical arrow keys.

ⓘ *The display is updated in real time if conditions change.*

4.4.7. Misc sub-menu



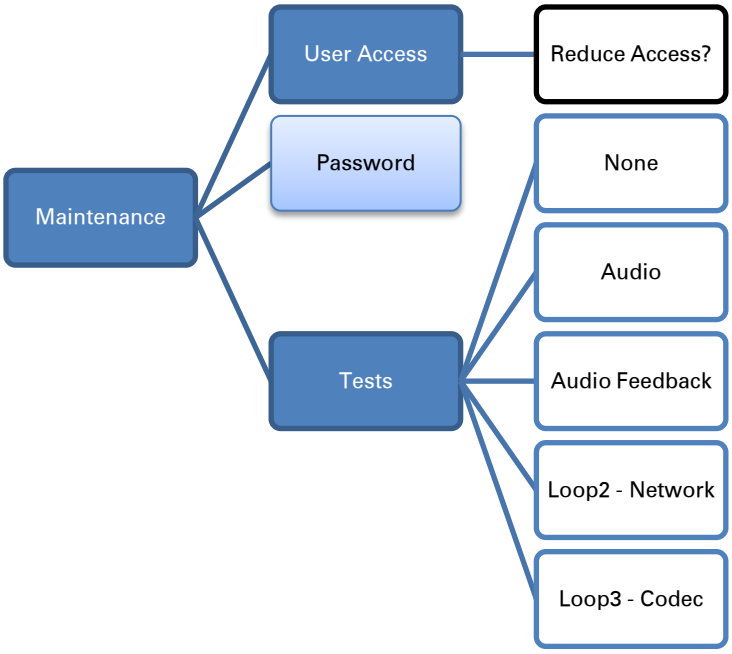
Notes:

- **General Reset**: you can select to delete only the snapshots or the profiles, or conversely to reset all parameters but keep these memories ("**Reset Settings**" choice). For a complete erasure with the restoration of all default settings, select "**Factory Reset**". *Caution: this cannot be cancelled!*
- **Aux. Functions**: this sub-menu allows to enable the auxiliary functions ((details on these functions can be found in 2.4, Auxiliary functions), and select the baud rate for the data channel. Besides, it also allows to enable or disable the displaying of the received SMS on the LCD (active by default).
- **Brightness**: select the preferred front panel display brightness
- **Language**: select the preferred language, and the menus are immediately updated.
- **5AS**: by default the 5AS system for ISDN (cf. page 9, 5A System®) is active. If required, it can be disabled using this sub-menu.
- **Auto Redial**: this sub-menu deals with the auto redial feature, whose operation is described in 3.5.2 above. For the double ISDN codec mode, auto redialling can be enabled separately for each codec.
- **Redial Attempts**: you can adjust from 1 to 20 the number of attempts to reset the link in case of a drop, or even "**Infinite**" for redialing indefinitely.
- **Time before dial**: (in seconds) adjust between 1 and 30 seconds, after a link loss, the waiting time before redialing. It is often worth leaving such a pause, so that the cause of the link loss is possibly cured, or the remote unit gets back to its normal state, etc.
- **Loop Control**: for activating the loop control feature (cf 2.5.5, Loop control and status). *Warning: this feature is incompatible with controlling links using the front panel keypad or the embedded html pages!*
- **LL Backup**: this sub-menu leads to settings for backing up a LL link using an IP or ISDN link (see 2.1.7 and 3.5.4, Setting up a backup link).
- **Backup Call**: this setting selects which interface is used to launch a backup call when the control loop is activated (see 3.5.4, Setting up a backup link).
- **Backup Receive**: enables the reception of an incoming IP or ISDN call when the unit is in LL mode. This parameter should be set on the "Receiver" codec for a backup link (see 3.5.4, Setting up a backup link). More generally, it can be used to accept incoming IP or ISDN calls even when the unit is on LL mode.

4.4.8. Maintenance sub-menu

On one hand this sub-menu allows managing the restricted access mode, whihc is specifically described further in 4.7, Restricted operation mode.

On the other hand it allows activating test loops (as described in 2.5.8, Test functions).



For setting up the restricted access, first you must set a password (that will be needed to later unlock the device), and for this use the "**Password**" sub-menu.

To activate the restricted mode, select the "**User Access**" sub-menu, and confirm by selecting "**Yes**". Then you must enter the password to lock the unit (this in order to make sure you know this password!). The unit immediately enters the restricted mode and will stay in this mode (even after a power cycle or reboot) until it is unlocked using the password.

If you lose the password: it is still possible to apply a complete factory reset; this will also clear the password and restore full access. The various methods are recalled in chapter 4.8 further, Clearing all settings.

4.5. Setting up a link

Two essential aspects have a major impact on the way that links can be managed with the SCOOP 5:

First, various types of networks are used for transmission. Two cases differ from the others:

- The transmission in « leased line » mode is permanent, without specific setup/release procedures.
- The transmission over the ISDN can use two channels simultaneously, or even more, and so imply dealing with two (or more) destination numbers.

Second, in ISDN mode the unit can also be operated as a "double codec" capable to transmit simultaneously two audio channels with 7 kHz bandwidth. This has an impact on the operating mode.

4.5.1. Setting up a link in ISDN mode

Preliminary settings

The network interface must be configured depending on the local ISDN line that is used, thanks to the "ISDN Parameter" sub-menu (**Setup** / **Network** / **ISDN Parameter**)

Protocol

The default setting is "Euro ISDN", also known as ETSI protocol. Change this setting if another protocol is needed in your location.

Local address

In some cases, it may be necessary to set the local address (or local ISDN number) of the line, and/or it is possible to assign a sub-address to the codec.

The local number allows "multiple subscriber numbering" or MSN. This number is usually the number remote equipment must dial to call your equipment. Configuring this number in the equipment is not mandatory if the equipment is directly connected to the public network. On the other hand, if the equipment is connected to a PABX, the number(s) are often required. The PABX may also impose a unique number for each B channel within the same BRI interface. In such a case, refer to the characteristics and configuration of the PABX.

ⓘ *Proper configuration of the local numbers is essential, and many problems in setting up links originate from mistakes or misunderstandings regarding this configuration. In doubt, leave this number blank! This is usually appropriate for public lines.*

Sub-address SA

This number differentiates several terminals connected to the same ISDN bus, which are allocated the same call number(s). Thus it can be useful in case other devices are connected with the SCOOP 5 on the same line.

Whenever a sub-address is set, the unit will only accept incoming calls specifically directed to this sub-address.

ⓘ *Most often, the best setting is to leave this blank, except in double codec mode where this setting is useful (see further the processing of calls in double codec mode).*

SPID

This "Service Profile Identifier" is only relevant for the NI-2 protocol (used by some operators in North America). This number identifies the services of the ISDN line, and usually includes the subscriber number or part of it. This number is mandatory to operate the line, and must be provided by the network operator.

Calling an ISDN number

To set the link, first set –if not already done- the unit in ISDN mode (menu **Setup** / **Network** / **Change Network** / **ISDN**). Then select the desired encoding format (**Setup** / **Algorithm**).

ⓘ *You can get a fast setting by simply selecting one of the "recently used" configurations displayed; otherwise select "**Other**" for a new coding configuration.*

Come back if needed to the base screen using Esc. Then enter the ISDN number of the destination and press the key.

If a sub-address is needed, right after the number enter the "*" character and the sub-address (4 digits max.). The number then has the form nnnn*ssss, e.g. 0912345678*32 (then press).

When the selected encoder needs two or more B channels, the unit asks for additional numbers. If the same number is suitable, just press the "green phone" button without re-entering a number.

An error message is displayed in case of a failure of the link establishment, or on a remote release. *To come back to the base screen, press* Esc *or* .

Receiving calls

When the 5A System is active, receiving calls is simple. When a call is received, the codec automatically "unhooks" and recognises the coding algorithm and protocol used, and finally sets the link automatically. On the receiving side, the unit will "follow" the calling unit.

When the 5A System is not active, you should first configure the codec for the desired coding algorithm and configuration. When a call is received, the unit will synchronise with the calling device, but the link will usually fail if the calling party has used another coding configuration than expected. *However, if J52 is used by both parties, the link will succeed even without 5AS active.*

ⓘ *Important notice: the unit can receive and accept incoming ISDN calls even when its current network interface is not ISDN, except if it is busy with an established link. On the other hand, it will ignore such calls if it is running in LL mode, except if the "Backup receive" mode is active (for details see above 3.5.4, Setting up a backup link).*

Releasing/Hanging up

In any case, to stop a link, press the key.

ⓘ *Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)*

Quick redialing

Once a number (or more for ISDN links with more than one B channel) has been called, it is easy to recall it without having to type it again: press the key, then you can scroll through the "history" (last dialled numbers) using the arrows. Press again the key when the desired number is displayed. This is especially efficient for quickly redialling the previous number.

4.5.2. Setting up links in double ISDN codec mode

The double codec mode is specific, as the unit behaves as two independent G711/G722 mono codecs, designated "Codec 1" and "Codec 2". Each codec sets or releases links separately on the B channel it is assigned, and uses the audio input and output it is assigned: input and output A (or "left") for Codec 1, and input and output B (or "right") for Codec 2.

Preliminary settings

The network interface must be configured depending on the local ISDN line that is used, just like in the normal ISDN mode (see above 4.5.1). However, in this mode, configuring a sub-address on each B channel (hence each codec) can be very useful. Indeed this is a means, when receiving calls, to differentiate the two codecs (which often answer the same subscriber number as they are on the same ISDN line). *But be aware these addresses cannot be dialed from a remote analog telephone.*

Besides, make sure to select the double codec mode: … / **ISDN Parameter** / **Codec Mode** / **Double Codec.**

Calling an ISDN number

To set the link, first set –if not already done- the unit in ISDN mode (menu **Setup** / **Network** / **Change Network** / **ISDN**).

Note that the display (base screen) has changed to show the respective status of the two codecs.

Then select for the relevant codec the desired encoding format (**Setup** / **Algorithm**): first select the codec (Codec 1 or Codec 2), then select the algorithm.

Come back if needed to the base screen using Esc. Then enter the ISDN number of the destination (followed by * + sub-address if needed) and press the key: you should then select the line/codec by pressing the L1 or L2 key depending on the codec you want to use.

An error message is displayed in case of a failure of the link establishment, or on a remote release. *To come back to the base screen, press* Esc *or* .

Receiving calls

Receiving a call works in the same way as in te single codec mode, except for the following:

- If the codecs are differentiated with exclusive subscriber numbers, a remote call can be sent specifically to one of the two codecs.
- You can have the same capability using the sub-addresses (*but this cannot be used by a remote analog phone*).
- Only G711 and G722 algorithms are supported by the codecs ion this mode.
- If the call is not specifically addressing one of the two codecs, the first available one (priority to codec 1) will pick up the call. Example: if codec 1 is busy, then codec 2 will pick up the call.

Releasing/Hanging up

To stop a link, press the key. You must then select which line/codec you want to release, using key L1 or L2. Press a second time to confirm.

Quick redialing

Once a number has been called with a codec, it is easy to recall it without having to type it again: press the key, then you can scroll through the "history" of calls using the arrows. Press key L1 or L2 depending on the desired codec.

4.5.3. Setting up a link in IP mode via Ethernet

A link is set up in a similar way as an ISDN link. The difference is mainly that instead of the telephone number, we use either an IP address, or a SIP URI (Uniform Resource Identifier).

Preliminary settings

If a SIP server is used, the SCOOP 5 must be registered on this server. Use the **AoIP Parameter** menu (*see above on page 51*) for entering the SIP account data and activate the SIP registration). *Check that the codec is successfully registered, using the **Tools** / **About** sub-menu: **SIP Status** information.*

Besides, for links over the Internet, most often the link passes through a NAT router which "hides" to the codec the characteristics of the public access: public IP address and RTP ports actually used. This can make the SIP transactions fail, or get a non-optimal route through the network. To avoid this a STUN server is often used; the corresponding settings are also found in the **AoIP Parameter** sub-menu.

*Check that the public address discovery has succeeded: **Tools** / **About** sub-menu: **Public IP** information (its mere presence shows the discovery has succeeded).*

ⓘ *You can find in annex 7.3 (Some methods to deal with NAT routers and firewalls) some additional information on the use of STUN.*

Directly call an IP address

This is the most basic way of setting the link. It is suitable only if:

- The other unit is "directly" reachable, i.e. there is no NAT Router or firewall blocking the connectivity. The simplest case is when both units are on the same LAN or private network.
- The IP address of the other unit is known!

To set the link, first set the unit in IP mode (**Setup** / **Network** / **Change Network** / **Ethernet**), and set the desired encoding format (**SETUP** / **Algorithm**, etc.).

ⓘ *Note that you can get a fast setting by simply selecting one of the "recently used" configurations displayed; otherwise select "**Other**" for a new configuration.*

Then enter the IP address and press the key.

ⓘ *When operating in this way, it is recommended to disable the SIP registration (**SIP Registration** parameter in the **AoIP Parameter** sub-menu, see on page 51).*

Calling via a SIP server

This is the technique when both units are registered on a SIP proxy server. In this case, each unit is identified by its SIP URI, in the form username@sipservername, like an email address. There is no need to know any IP address (and hence there is no problem if the IP address of a unit changes for whatever reason).

To set the link, first set the unit in IP mode (**Setup** / **Network** / **Change Network** / **Ethernet**), and set the desired encoding format (**SETUP** / **Algorithm**, etc.).

Then enter[1] the SIP URI of the unit to call, and press the key.

ⓘ *It is often possible to dial the short form username (omitting the @sipservername) when the device is itself registered onto the same "sipservername" server.*

[1] Use the "#" key to switch the keypad mode numeric/lowercase/uppercase

Receiving calls

This is very simple, in both cases (direct peer to peer link or SIP server). There is nothing to do…

When a call is received, the units negotiate automatically a commonly acceptable coding algorithm, and set the link automatically. On the receiving side, SCOOP 5 will "follow" the calling unit.

ⓘ *Important notice: the unit can receive and accept incoming IP calls even when its current network interface is not Ethernet/IP, provided it not already busy with an established link. On the other hand, it will ignore such calls if it is running in LL mode, except if the "Backup receive" mode is active (for details see above 3.5.4, Setting up a backup link).*

"Network quality" setting

Depending on the quality of service provided on the network, especially its jitter performance, it is possible to change the stability/latency compromise used by the SCOOP 5. For this purpose, a setting is available in the menu (**Setup** / **Network** / **Ethernet Parameter** / **Network Quality**). Three choices are proposed:

- "HIGH": suitable for a good quality and low jitter network; latency is minimal, but the codec will have little tolerance to possible jitter
- "MIDDLE": intermediate (and default) setting, suitable for a moderate transmission jitter
- "LOW": to be preferred when the network has low QoS, especially for residential ADSL lines. This setting ensures a safer operation, at the cost of a high latency.

On a LAN and/or private network with a controlled quality, the "HIGH" quality setting is recommended, as it yields minimum latency.

On the contrary, it should be avoided for a link via the Internet, as it can only tolerate a low jitter. One solution can be to start with a "MIDDLE" setting, and switch to the "LOW" setting if too much audio disturbance is heard.

Quick redialing

Once a remote device has already been called, it is easy to recall it: press the key, then you can scroll through the call "history" using the arrows. Press again the key when the desired number is highlighted. This is especially efficient for quickly redialling the last called number.

Releasing/Hanging up

To stop a link, press the key.

ⓘ *Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)*

Links with IP phones

SCOOP 5 is compatible with IP phones that use the SIP protocol (many on the market do). The algorithm used in this case is commonly G711, but a few IP phones also accept G722.

Note that "IP phones" include software SIP phones implemented on computers (also called "softphones") or smartphones.

Multicast mode transmission

On a network that can support it[1], the multicast mode can optimise the resource usage when an audio stream has to be distributed simultaneously to several destinations. Contrary to the normal bidirectional unicast mode, this mode is unidirectional: one *sender* encoder sends a stream towards a multicast *group address*, and one or several decoders receiving the stream pick up the packets sent to this group address and decode the audio stream.

On the SCOOP 5 the operating mode stays quite similar to the "normal" mode, with mainly two differences in the multicast mode:

- A codec must be set as sender or receiver device
- SIP is not used and hence the SIP configuration is not relevant

For the operation it is assumed the network "statically" supports UDP multicast, i.e. routers of the network recognise and deal with routing the packets with multicast group addresses.

For obvious reasons the coding setup is entirely decided at the source. On the codec on the audio source side (hence *sender* of the encoded stream), the procedure for setting up a multicast stream is the following:

- Set the codec in "multicast send" mode. *To do this, refer to the* ***AoIP Parameter*** *sub-menu, page 51.*
- Select on this codec the desired audio coding parameters;
- Start the streaming as for setting up a call in normal mode: enter the IP group address, then press the key.

Note that the codec decodes its own stream on its audio output; this can be useful for audio monitoring.

For each codec that has to decode the stream, the procedure is simple:

- Set the codec in "multicast receive" mode. *To do this, refer to the* ***AoIP Parameter*** *sub-menu, page 51.*
- Start the decoder as for setting up a normal call: enter the IP group address, then press the key.

[1] This does not include the Internet; multicast cannot be used over the Internet.

4.5.4. Setting up a POTS link

Preliminary settings

The network interface must be configured depending on the PSTN line that is used, thanks to the "POTS Parameter" sub-menu (**Setup** / **Network** / **POTS Parameter**)

ⓘ *You should preferably disable services such as call waiting tones. Tone signals injected on the line during the communication have a high risk of disturbing the codec operation, and even dropping the link. It is not recommended to use a line if it is not possible to disable such tones.*

Calling a POTS number

To set the link, first set –if not already done- the unit in POTS mode (menu **Setup** / **Network** / **Change Network** / **POTS**). Then select the desired encoding format (**Setup** / **Algorithm**). Select the desired variation:

- P0: default mode of operation for this
- P1: protected mode 1, with increased resilience to errors (but higher latency)
- P2: protected mode 2, with slightly higher resilience to errors (and slightly higher latency)

ⓘ *Make sure to set units the same on both sides of the link, because this setting cannot be negotiated automatically by the units. It is recommended to use the default P0 mode, the other are mainly for compatibility with older codecs of the AETA product range.*

Come back if needed to the base screen using Esc. Then enter the number of the destination and press the key. The codec dials, negotiates with the remote unit the bit rate (this phase can last several seconds) and sets up the link and the coding.

ⓘ *A message is displayed in case of a failure of the link establishment, or on a remote release. To come back to the base screen, press* Esc *or* .

Receiving a POTS call

Receiving calls is simple. When a call is received, the SCOOP 5 "unhooks" and the link is set up after the negotiation of the bit rate with the calling unit (this can last several seconds).

ⓘ *Important notice: the unit can receive and accept incoming POTS calls even when its current network interface is not the POTS line, provided it is not already busy with an established link. On the other hand, it will ignore such calls if it is running in LL mode, except if the "Backup receive" mode is active (for details see above 3.5.4, Setting up a backup link).*

Releasing/Hanging up

To stop a link, press the key.

ⓘ *Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)*

Quick redialing

Once a remote device has already been called, it is easy to recall it: press the key, then you can scroll through the call "history" using the arrows. Press again the key when the desired number is highlighted. This is especially efficient for quickly redialling the last called number.

4.5.5. Setting up a link over a leased line

There is no specific procedure for establishing or interrupting such link: once the physical connections and the coding configuration are done, set the unit in LL mode (**Setup** / **Network** / **Change Network** / **LL**). The link is immediately started: the encoder transmits the encoded stream, and the decoder looks for synchronisation on the received stream.

In case the link is broken, the decoder restarts looking for synchronisation and restarts as soon as the stream comes back, without any action needed from an operator. This is also true when restarting after a power cycle.

By default, no incoming call from another network interface (Ethernet, ISDN, POTS...) is accepted when the unit is set in this LL mode. However, it is possible to allow such incoming calls by changing the "Backup Receive" setting (in the sub-menu **Tools** / **Misc** / **LL Backup** / **Backup Receive**, or the "Misc" tab in the html pages).

4.5.6. Setting up a mobile telephone link

Prerequisites

You must have a subscription and a SIM card granting access to mobile transmission. *In the case described here the subscription must include access to the basic telephone service.*

Preliminary settings

First insert the SIM card in the SCOOP 5 (the unit must be powered off for this), and connect at least one antenna to the unit. See how to proceed on page 32, "Drawer for SIM card [6]". Power up the codec, enter the sub-menu **Setup** / **Network** / **Mobile (Int.)** / **Mobile Parameter** / **PIN**. Enter the PIN code of the SIM card using the keypad and press OK. *This is useless, of course, if the card's PIN code is blank or disabled.*
The network access must be configured using the "Mobile Parameter" menu (**Setup** / **Network** / **Mobile Parameter**), see details in page 56, Mobile Parameter.
.

- Select the mobile phone mode: **Mode** / **Cellphone**.
- If needed select the network type: **Network Mode**. In doubt, select "**Auto**" which fits most situations.
- The **Network Operator** sub-menu allows you to choose among the available operators, if your mobile subscription allows you to do so.
- *Other settings (Network Quality, Access Point…) are meaningless for a telephone link.*

Calling a mobile number

To set the link, first set –if not already done- the unit in mobile access mode (menu **Setup** / **Network** / **Change Network** / **Mobile (Int.**). If needed come back to the base screen, using Esc.

ⓘ *Note that the display now shows you the network type (example: 3G) and the radio reception quality. Press* Esc *(display the root menu) and you get more details: operator name, more specific technology (example: HSPA).*

Enter the destination number and press .

An error message is displayed in case of a failure of the link establishment, or on a remote release. *To come back to the base screen, press* Esc *or* .

Receiving a mobile phone call

Receiving calls is simple. When a call is received, the SCOOP 5 "unhooks" and the link is set up.

ⓘ *Important notice: the unit can receive and accept incoming mobile phone calls even when its current network interface is not the mobile network, provided it not already busy with an established link. On the other hand, it will ignore such calls if it is running in LL mode, except if the "Backup receive" mode is active (for details see above 3.5.4, Setting up a backup link).*

Releasing/Hanging up

To stop a link, press the key.

ⓘ *Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)*

Quick redialing

Once a remote device has already been called, it is easy to recall it: press the key, then you can scroll through the call “history” using the arrows. Press again the key when the desired number is highlighted. This is especially efficient for quickly redialling the last called number.

4.5.7. Setting up a link in IP mode via a mobile network

Prerequisites

You must have a subscription and a SIM card granting access to mobile transmission. *In the case described here the subscription must include access to packet switched data transmission.* It is possible to operate over a 2G network in EDGE mode, if using very low bit rates, but for an acceptable performance it is preferred to access a 3G/3G+ network, at minimum UMTS.

Preliminary settings

First insert the SIM card in the SCOOP 5 (the unit must be powered off for this), and connect at least one antenna to the unit. See how to proceed on page 32, "Drawer for SIM card [6]". Power up the codec, enter the sub-menu **Setup** / **Network** / **Mobile (Int.)** / **Mobile Parameter** / **PIN**. Enter the PIN code of the SIM card using the keypad and press OK. *This is useless, of course, if the card's PIN code is blank or disabled.*

ⓘ *If you use a mobile access USB module, insert the SIM card in this device (refer if needed to its dedicated literature) and plug the USB key in the back of the unit. After a while, the module is detected and appears in the list of the network accesses under the name "Mobile 1" or "Mobile 1 (Ext.)". In all the instructions in this chapter, replace "Mobile (Int.)" by this name allocated to the USB device.*

The network access must be configured using the "Mobile Parameter" menu (**Setup** / **Network** / **Mobile Parameter**[1]), see details in page 56, Mobile Parameter.

- Select the packet data mode: **Mode** / **IP Mode**.
- If needed select the network type: **Network Mode**. In doubt, select "**Auto**" which fits most situations. *Avoid giving the priority to GSM, as its performance is poor.*
- The **Network Operator** sub-menu allows you to choose among the available operators, if your mobile subscription allows you to do so.
- Go to the sub-menu **Access Point** / **APN**: enter the APN code of the operator using the keypad, then OK.
- If required enter the other access point settings: user name and password. *This is n ot needed usually, otherwise the needed data are provided along with the subscription and the SIM card.*
- You can define a **Network Quality**: in doubt, select **Middle**. *You can test a setting on a connection with a significant duration, and change the setting if the stability does not look sufficient.*

If a SIP server is used, the SCOOP 5 must be registered on this server. Use the **AoIP Parameter** menu (*see above on page 51*) for entering the SIP account data and activate the SIP registration. *These settings are common to IP modes both via Ethernet and mobile networks.*

Besides, for links over the Internet, most often the link passes through a NAT router which "hides" to the codec the characteristics of the public access: public IP address and RTP ports actually used. This can make the SIP transactions fail, or get a non-optimal route through the network. To avoid this a STUN server is often used; the corresponding settings are also found in the **AoIP Parameter** sub-menu. *These settings are common to IP modes both via Ethernet and mobile networks.*

ⓘ *You can find in annex 7.3 (Some methods to deal with NAT routers and firewalls) some additional information on the use of STUN.*

[1] Or "Mobile 1" for an external USB device

Calling via a mobile IP network

To set the link, first set –if not already done- the unit in mobile access mode (menu **Setup** / **Network** / **Change Network** / **Mobile (xxx)**[1]. If needed come back to the base screen, using Esc.

ⓘ *Note that the display now shows you the network type (example: 3G) and the radio reception quality. Press* Esc *(display the root menu) and you get more details: operator name, more specific technology (example: HSPA).*

Enter the destination number and press .

From this moment you can set a call just like a call over the Ethernet interface (see above in *4.5.3, Setting up a link in IP mode via* Ethernet):

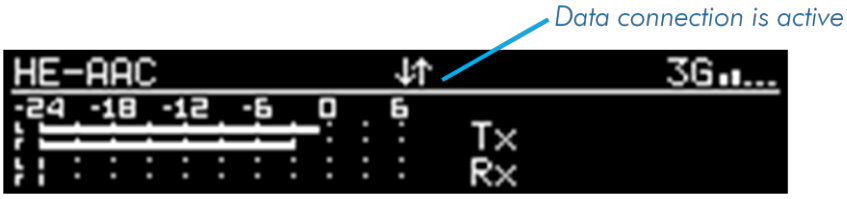
- Call directly, without a SIP server: enter the remote device IP address and press the key.
- Use a SIP server: enter the SIP URI of the remote device and press the key.

An error message is displayed in case of a failure of the link establishment, or on a remote release. *To come back to the base screen, press* Esc *or* .

Receiving a mobile IP call

Receiving a call is basically the same as through the Ethernet interface, but with some significant differences:

- SCOOP 5 must be connected to the mobile data service to be able to receive requests from the network (such connection is somewhat equivalent to the physical connection of an Ethernet cable). This connection is performed automatically by SCOOP 5 just before setting an outgoing call. To make it "manually" and get ready to receive a call, you must use the sub-menu **Setup** / **Network** / **Mobile Parameter / Data Connection**. *The "Connected" status is recalled on the base screen and the main menu*:



- Mobile networks often include devices blocking the access *towards* a mobile terminal: NAT routers, often symmetrical (which prevents a STUN server to be efficient), etc. Moreover, you rarely get a public IP address, and it is never possible to get a public and *fixed* IP address.

For these reasons it is rather difficult to receive a call without using a SIP server. Conversely, if the SCOOP 5 is registered on a SIP server, once the data connection is active the codec is actually able to receive a call.

When a call is received, the units negotiate automatically a commonly acceptable coding algorithm, and set the link automatically. On the receiving side, SCOOP 5 will "follow" the calling unit.

ⓘ *Important notice: the unit can receive and accept incoming IP calls even when its current network interface is not mobile IP, provided it is not already busy with an established link. On the other hand, it will ignore such calls if it is running in LL mode, except if the "Backup receive" mode is active (for details see above 3.5.4, Setting up a backup link).*

Releasing/Hanging up

To stop a link, press the key.

[1] The actual name of the network access depends on the concerned module: "Mobile (Int.)" for the module integrated with the "wireless" option, "Mobile 1 (Ext.)" for an external USB device, etc.

Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)

Quick redialing

Once a remote device has already been called, it is easy to recall it: press the key, then you can scroll through the call "history" using the arrows. Press again the key when the desired number is highlighted. This is especially efficient for quickly redialling the last called number.

4.6. Management of the configuration profiles

The configuration profiles ease configuration changes and the setting up of routine links. There are three types of profiles:

- "**Remote profiles**", which include call parameters for a given **destination** (number(s), coding, etc.). Such a profile is similar to a directory entry, but in addition it can memorize the coding parameters.
- "**Local profiles**" memorize the **network access** parameters. These are the settings found in the sub-menu **Setup** / **Network**. Recalling a local profile is a quick way to restore the configuration needed for a given line or network.
- "**Snapshots**" memorize the settings for the **audio** interfaces. These are settings found in the **Setup** / **Audio** sub-menu.

These various memories are usable locally, but also through the embedded html pages, and also using these pages they can be imported/exported from/to a computer.

Each remote profile includes the following elements:

- *Name* assigned to the profile
- *Number(s)*: ISDN, POTS or mobile call number(s), or IP URI, Ip address, depending on the transmission mode.
- *Coding configuration* associated with the profile

A remote profile may have no number included; in such case it is used for quick and safe recall of a given coding configuration.
Conversely a remote profile may have no associated coding configuration included; in such case it is just like a directory entry, used for instant dialling a known destination.

The profiles are managed and used via the "Profiles" menu (refer to details in 4.4.4, **Profiles** menu). For remote profiles, the available possibilities are:

- Create a *New* profile, assigning it a name, possibly call number(s), and possibly an associated coding configuration.
- *Load* a profile previously recorded: the codec then configures itself with the coding configuration memorized in the profile.
- *Call* directly a profile: the codec then configures itself with the coding configuration memorized in the profile (if any), and sets a call to the number(s) included in the profile.
- *Delete* a profile previously recorded.
- *Update* profile previously recorded, either for renaming it or change its content.

For local profiles or snapshots, the available possibilities are:

- Create a *New* profile, which will memorize all the current network settings (for a local profile) or all the audio settings (for a snapshot), assigning it a name.
- *Load* a local profile or snapshot: the codec then restores the settings included in the profile.
- *Delete* a local profile or snapshot.

4.7. Restricted operation mode

4.7.1. Principles

SCOOP 5 can be locked in a restricted user access mode, which limits user access to the essential controls and features. For routine operation, this mode allows to avoid risks such as random settings, with unpredictable effects, handling or adjustment mistakes, accidental erasure of useful memories, etc.

When SCOOP 5 is in restricted access, it is still quite operative, especially if suitable memories (profiles, snapshots) have been prepared beforehand, because the following functions are still accessible:

- Receiving calls
- Releasing links
- Redialing (recalling previous numbers) outgoing calls
- Loading and calling remote profiles
- Loading local profiles or snapshots

On the other hand, it is not possible to change settings via the menus, as only the "Profiles" menu is accessible. It is also impossible to delete or edit the recorded profiles.

A password is needed for switching to the restricted mode (locking) or back to full access (unlocking). This password can be programmed (of course, not when in restricted access...).

Warning: this access protection is completely independent from the one that can be set for the "web" interface (embedded html pages). The passwords have no relationship, and are set separately. One of the interfaces (front panel or html) can be locked while the other is not.

4.7.2. Locking the front panel

Managing the restricted mode is done with the sub-menu **Tools** / **Maintenance**. See in 4.4.8 (**Maintenance** sub-menu) this sub-menu items.

First you must program the desired password, because the password is blank when the unit comes from the factory or after a complete "factory reset". The password is an arbitrary sequence of numbers.

For switching to the restricted mode, select **User Access** and confirm (**Yes**). You must then enter the password to actually lock the unit (this makes sure you really know the password before locking). The screen warns you about the switching to restricted access.

Press Esc and come back to the base screen. From then on, on every attempt to navigate in the non authorized menus, the following message is displayed:

4.7.3. Unlocking the front panel

From the base screen or the root menu, type the password sequence: SCOOP 5 unlocks:

The full access is restored. Press Esc to come back to the base screen.

4.7.4. Lost the password?

You have forgotten or lost the password ? The only way to restore full access is to clear all the settings, which will also clear the passwords. The unit will be accessible again.

Chapter 4.8 below shows the procedure for such erasure.

ⓘ *In order not to lose all the memorized settings, you can perform a complete reset of the settings without deleting the profiles. These can be useful for you to come back more easily to the desired configuration.*

4.8. Clearing all settings

In some cases (especially if the password is lost), it can be necessary to come back to the "factory" configuration.

To clear up all the configuration and come back to the factory settings, normally you should go to the sub-menu **Tools** / **Misc** / **General Reset** and select **Factory Reset**, or **Reset Settings** (which does not delete the memorized profiles and snapshots).

The default password is blank. *In the factory setting, menus use the English language. Other languages can be selected via* ***Tools*** / ***Misc*** / ***Language.***

If it is not possible to access the menu to perform the factory reset (password lost), the erasure should be done using the embedded html pages: see chapter 5 below for the operating mode.

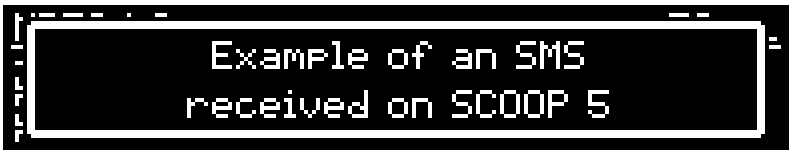
4.9. Backing up and restoring the configuration

It is possible to save all the equipoment settings ina file, and conversely to restore a comple configuration froma file previously recorded in such way. Such transfers can be done using the embedded html server; refer to 5.10.6 for the operating mode.

Partial exports are also possible, selecting eleme,ts to include in the export: remote profiles, local profiles, snapshots, settings…

4.10. Displaying received SMS

When the unit is equipped with the "wireless" option and it is registered on a mobile network, SCOOP 5 can display SMS received on the SIM card's subscriber number. The operation is very simple: on receiving the SMS the text is directly displayed on the screen:



Whenever the complete message cannot be displayed on the screen, use the arrows to scroll the message and read all the text. After the message is read, press Esc to come back to the normal display.

Caution: the message is now cleared and it cannot be displayed again.

If you do not want to display the received SMS, you can disable this feature: go to the sub-menu ***Tools*** / ***Misc*** / ***Aux. Functions*** *and disable the* ***SMS*** *parameter.*

5. Operating mode – Embedded HTML pages

The embedded html server in SCOOP 5 provides a comfortable and efficient means to control and monitor the unit. It just needs the SCOOP 5 to be connected to an IP network and to be reachable from a computer, or another device with an html browser: tablet, smartphone…

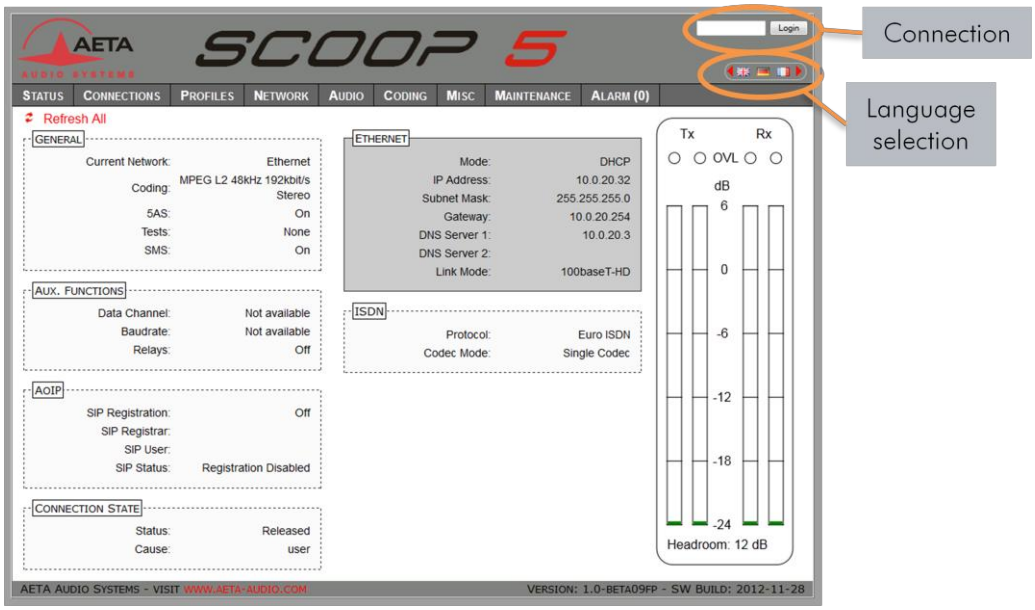
In the most common case, the two devices are connected on the same local network (LAN). But it is also possible to control the SCOOP 5 by remote, provided that the control device can reach it (TCP/IP port 80, HTTP protocol).

If you also have a secondary Ethernet interface (using a USB/Ethernet adapter), each of the two interfaces can be used for this remote control access.

This control mode is usable regardless of the OS of the control unit, and the embedded pages are compatible with all common browsers. No software installation is needed on the control position.

5.1. Accessing the SCOOP 5 html pages

Once the SCOOP 5 is connected on an IP network, the first step is to get its IP address, from the menu: **Tools** / **About** / **Local IP**[1]. Then, on the control device, launch the html browser and enter the IP address of the SCOOP 5 in the "address" or "URL" field. This gives access to the html server integrated in SCOOP 5. The page which is displayed is similar to the following picture:



If needed, select another language by clicking the suitable flag (*this choice is not linked to the language selected for the menus on the front panel interface*).

The home page displayed above is the "**Status**" page, which provides an overview of the unit status, but allows no action on it. This is the only "free access" page, with no limitation or access control.

To access the other pages, you must "log in", and get for the control device an exclusive access. Any connection request from another device will remove this access.

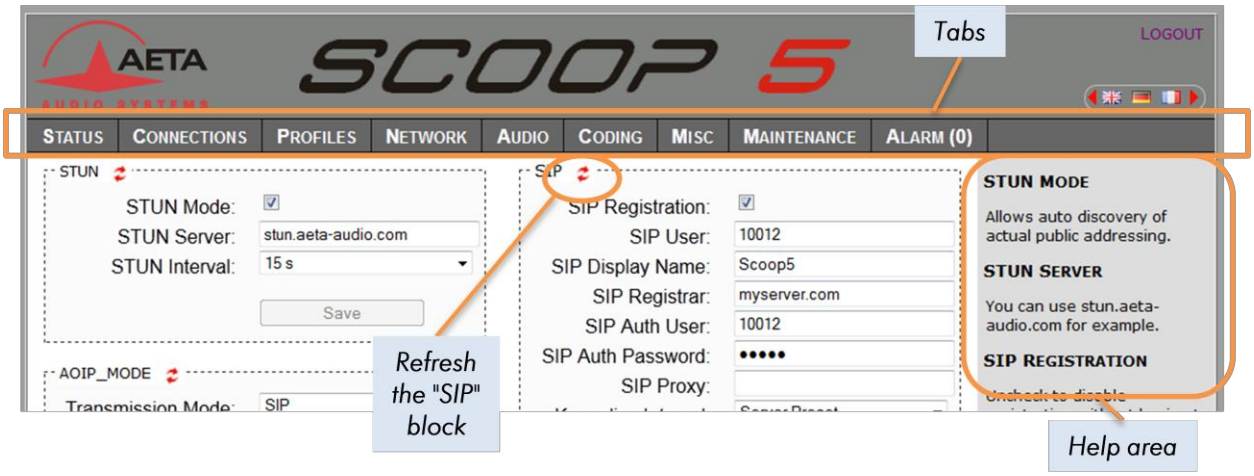
To log in, enter the password and click the connection button. The initial password is blank: click directly on the button. To set the password and enable protection, go to the "Maintenance" page (cf. further).

To release control, click on "Logout" (also in the connection area). You are also logged out automatically after a long period of time with no action on the pages.

[1] For a possible secondary Ethernet interface, menu Config / Network / Ethernet 1 Parameter...

5.2. Principles of operation with html pages

The picture below shows a typical page.



On top you find a bar of tabs corresponding to the various categories of functions and parameters for the SCOOP 5. Clicking a tab you access either a page, or a drop-down list for selecting a secondary page. *These tabs and pages are detailed in the following chapters.*

Under the bar can be found the information and adjustable parameters, with various selection or entry modes for these parameters, grouped in blocks (each surrounded with a frame). On the right side, a text area provides additional help and hints.

As a general rule, the displayed parameters are read when accessing the page, and are not refreshed automatically[1]. To force a refresh, click the icon: the data in the area or frame are read and refreshed.

Exception: some data on some pages is however periodically and automatically refreshed. This makes a modest bit rate, but you should exit the html pages if you want no traffic at all on the path between SCOOP 5 and the control device.

Two tabs have a specific behavior:

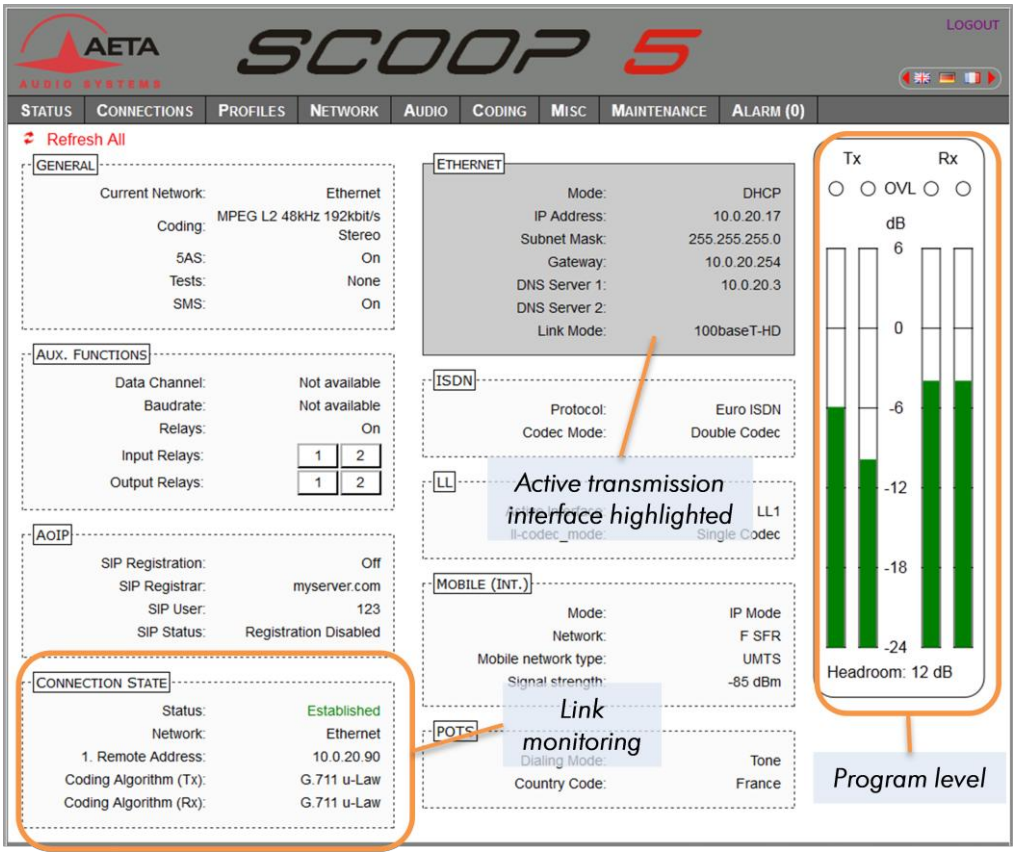
- "Status" is accessible without a login and some data are updated automatically.
- "Alarms" is also updated automatically, and switches to red when an alarm triggers, showing the number of issues detected. You can then check for details by clicking the tab.

Access to the tabs (other than "Status") requires loggin in beforehand. If you click a tab without being logged in, the login dialog box opens to allow you to enter the connection password. If it is blank, just click the "Login" button.

[1] This is on purpose, in order to avoid a permanent high rate of queries to the unit, which may be questionable in some situations.

5.3. "Status" tab

This tab leads to the "Status" page, which is also the home page. It provides an overview of the essential settings and parameters, sufficient for basic monitoring.



This page is accessible without a log in, and without a password.

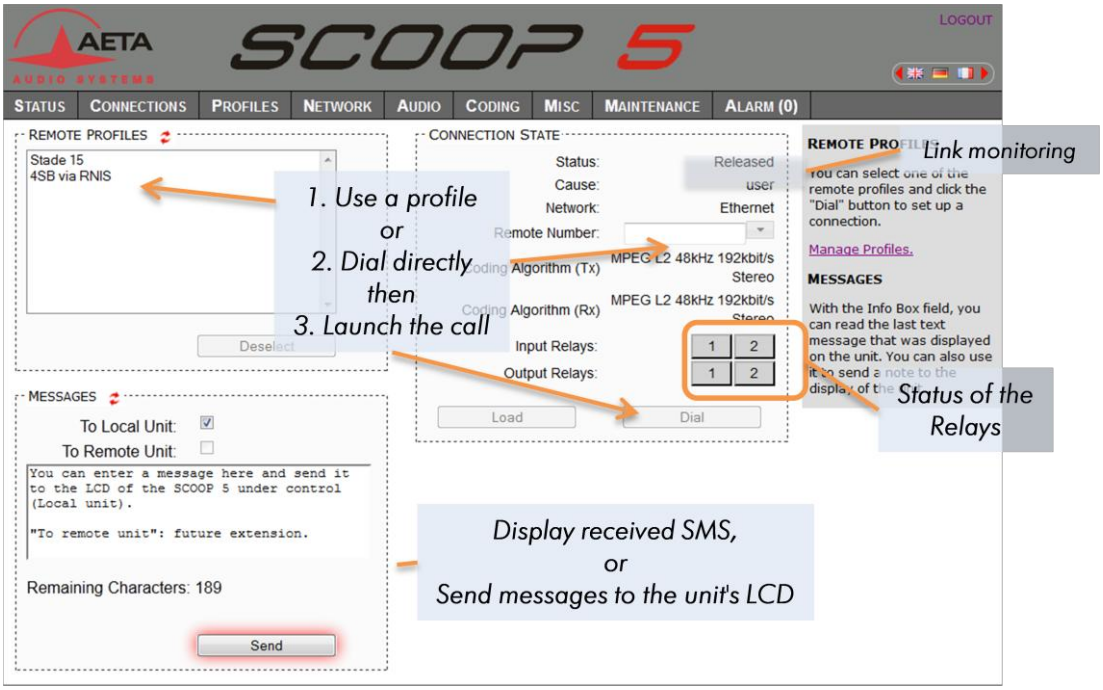
The link status is monitored dynamically; e.g. you can see call reception and releasing. It is also possible to monitor calls that an operator is managing using the front panel.

The page displays the audio level of the two transmitted signals (encoder) and the two received signals (decoder), shown as bargraphs with a 0 dB reference. *Refer to chapter 4.4.3 (Audio sub-menu) for more details on the setting for this reference. The "Headroom" setting is adjustable using the "Audio" tab.*

ⓘ *Note: the goal of these bargraphs is to provide an indication on the presence and level of the audio modulation. The measurement has a good precision for stationary signals (1 dB accuracy and resolution), but the update rate is rather slow. As a consequence these bargraphs are not appropriate for precisely monitoring a dynamic program, and they should not be considered an alternative to real program meters!*

5.4. "Connections" tab

This tab leads to the "Connections" tab, allowing to remote control the links: monitoring, outgoing call set up and call releasing.



The "Connection State" block allows to monitor and/or manage links.

- For direct dialing: the codec has to be set beforehand for the right interface (see "Network" tab), and the suitable coding configuration ("Coding" tab). Enter the number (or numbers, or SIP URI…), then click "Dial". *You can also use one of the last called numbers: click the arrow on the right of the "Number" field and pick a number in the dropdown list.*
- For using a remote profile: select the profile in the list on the left, and click the "Dial" button. Alternatively, you can click "Load". The codec loads the profile settings but does not dial: this is useful for quickly setting the coding parameters.

Indicators show the status of the relays when the "Relay transmission" feature is active. *For auxiliary functions, see the "Coding" tab.*

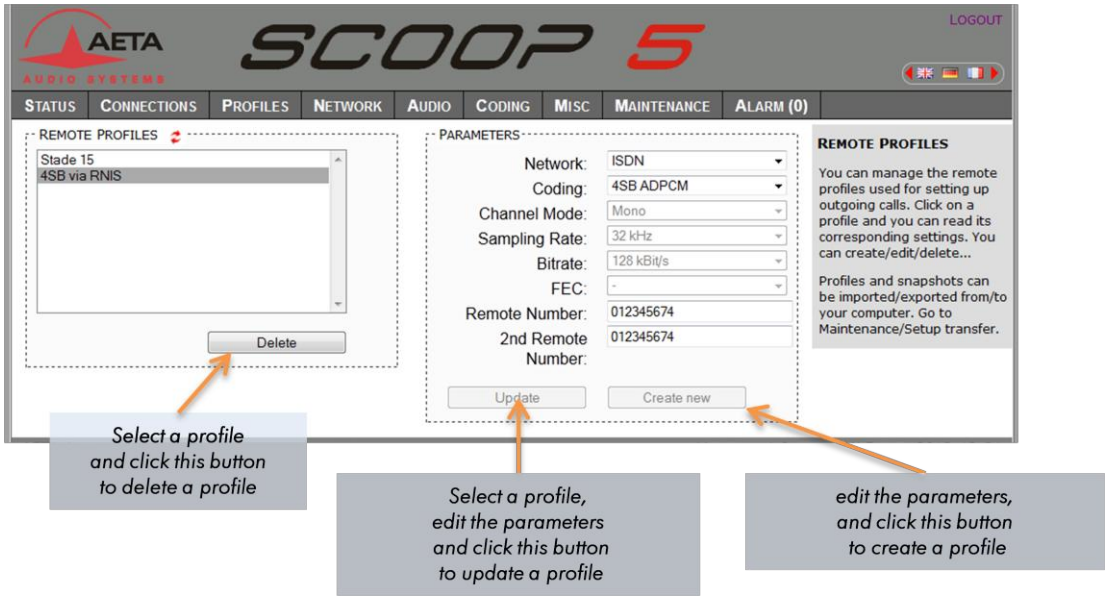
The SMS received by SCOOP 5 are displayed in the "Messages" frame (in addition to being displayed on the unit's LCD). This block can be used also to send a text to the LCD from the control position.

ⓘ *When SCOOP 5 is in double ISDN codec mode, the "Connection State" frame is replicated, one frame for each codec. But selecting a remote profile switches back to the normal display (for the single codec), because the profile data correspond to a single codec (remote profiles cannot be used in double codec mode).*

5.5. "Profiles" tab

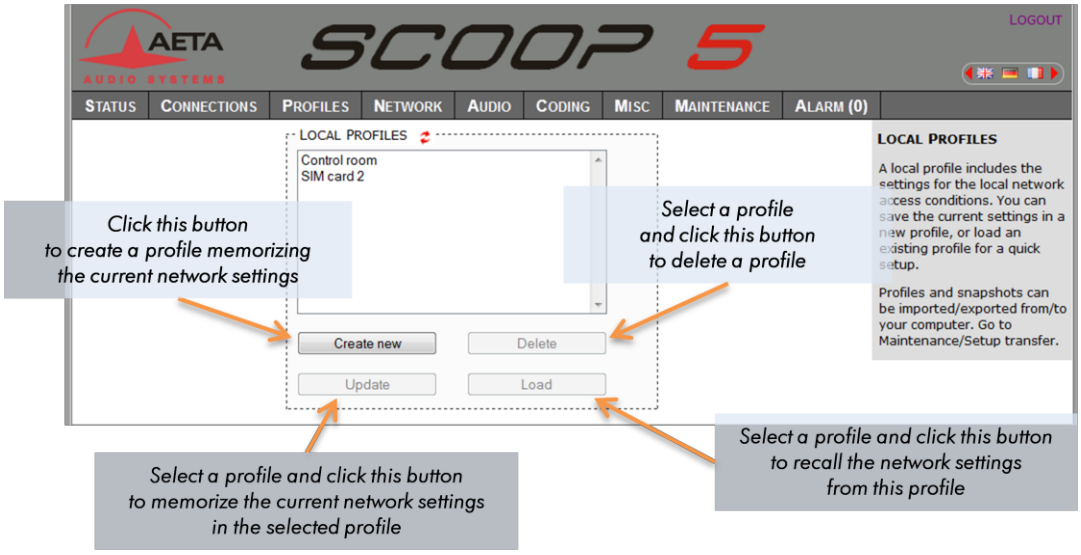
This tab proposes three secondary pages, one for each profile type.

5.5.1. "Remote profiles" page



Parameters that are not relevant in a given context are shaded and inactive.

5.5.2. "Local profiles" page


AETA
AUDIO SYSTEMS
SCOOP 5
LOGOUT
STATUS
CONNECTIONS
PROFILES
NETWORK
AUDIO
CODING
MISC
MAINTENANCE
ALARM (0)
LOCAL PROFILES
Control room
SIM card 2
Create new
Delete
Update
Load
LOCAL PROFILES
A local profile includes the settings for the local network access conditions. You can save the current settings in a new profile, or load an existing profile for a quick setup.
Profiles and snapshots can be imported/exported from/to your computer. Go to Maintenance/Setup transfer.
Click this button to create a profile memorizing the current network settings
Select a profile and click this button to delete a profile
Select a profile and click this button to memorize the current network settings in the selected profile
Select a profile and click this button to recall the network settings from this profile

5.5.3. Snapshots

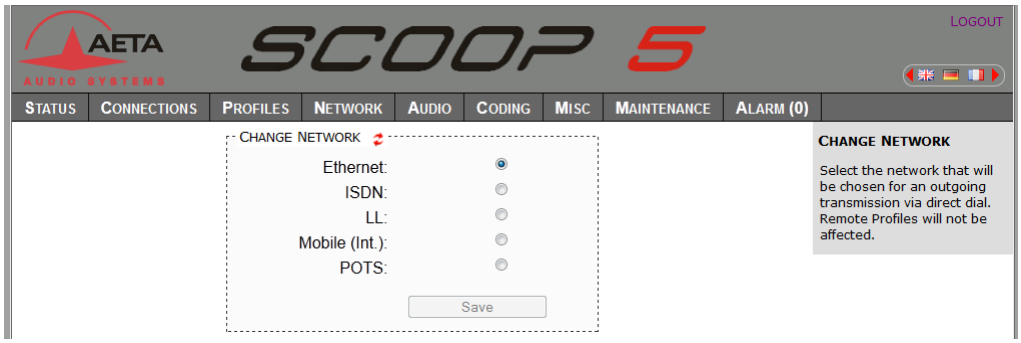
Snapshots are in fact managed on the "**Audio**" page, and this "**Snapshots**" choice on the "**Profiles**" tab just redirects to the "**Audio**" page. *See further the description of the audio page.*

5.6. "Network" tab

This tab leads to several pages, which correspond, just like the sub-menu "**Network**" of the front panel interface (see 4.4.1 above), to the various network interfaces. For this reason, the number of pages actually proposed depends on the version and the options of the unit (with or without ISDN, mobile access, etc.).

5.6.1. "Change Network" page: default interface selection

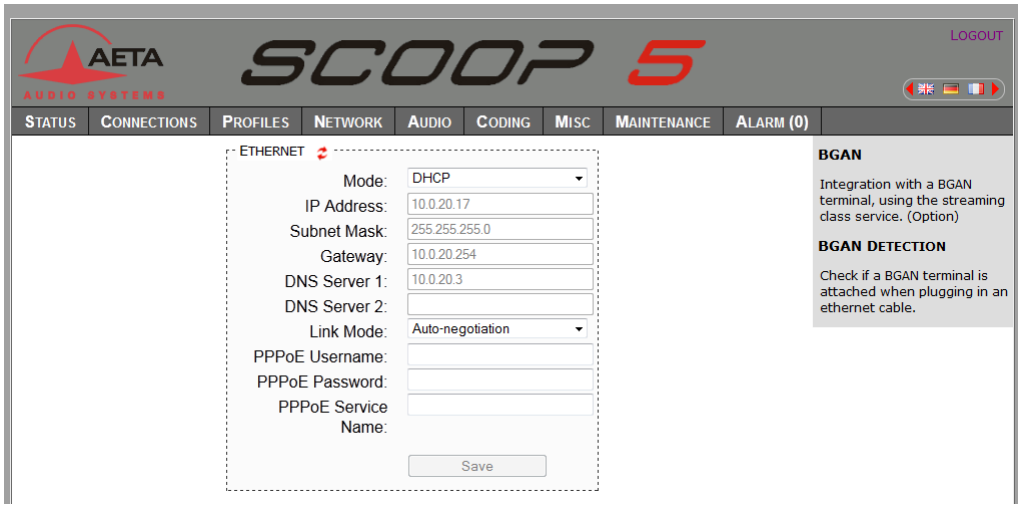
This page is used for selecting the default interface: this is the one that is implicitly used when dialing to set an outgoing call. Besides, codec algorithm settings (either via menus or the html pages) are applicable to this interface[1].



Select the desired interface and click "Save".

5.6.2. "Ethernet Parameter" page

More details on these parameters in 4.4.1, Ethernet Parameters.



Remember to **save the changes**, clicking the "Save" button!

Settings regarding the Ethernet interface and IP addressing are especially critical, because this is just the interface you are controlling SCOOP 5 through. You can lose control because of a mistake in the settings; it might even be the normal expected consequence for a configuration change. So be very careful with the settings on this page!

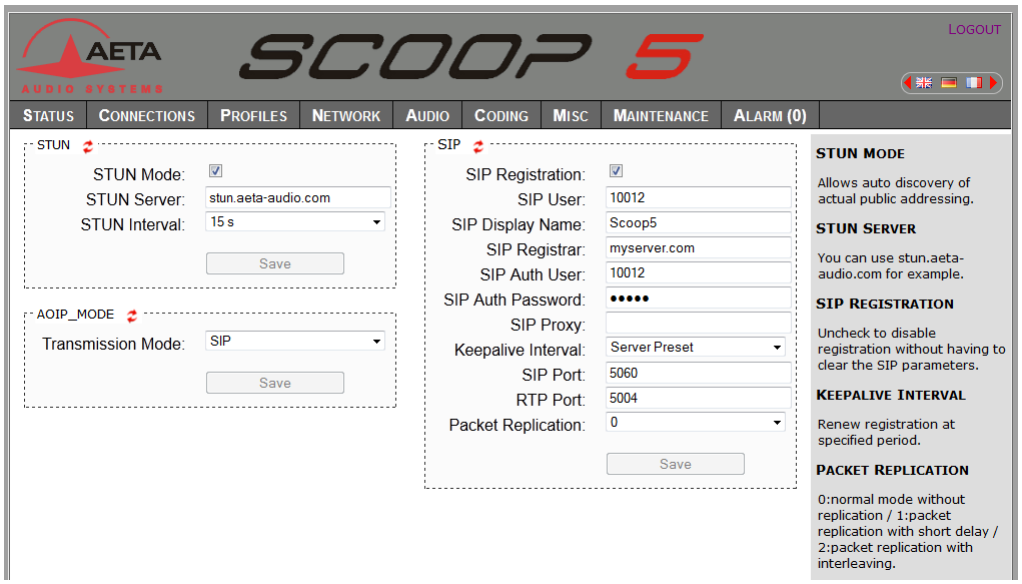
In case control is lost after an error, you must regain control and correct the settings using the front panel of SCOOP 5.

Note: if a secondary Ethernet interface is available, it is also shown in this page, in an "Ethernet 1" frame.

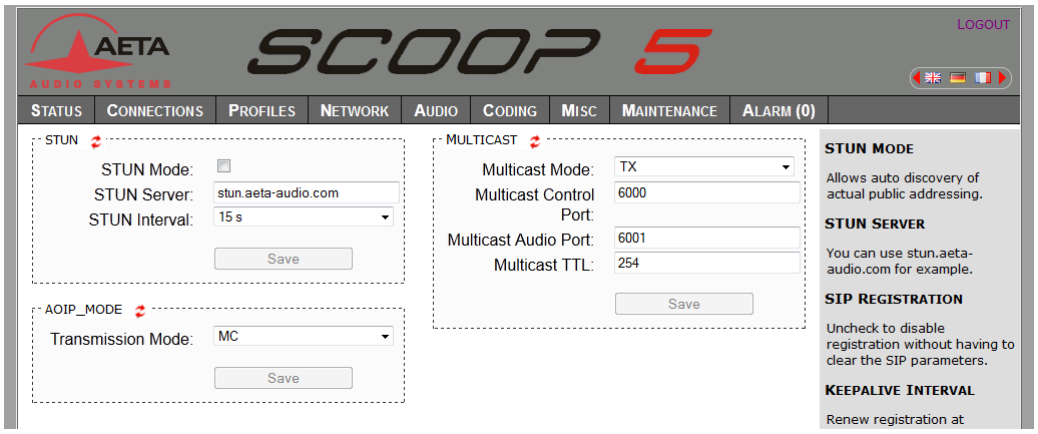
[1] Remind that the algorithm settings are memorized separately for each network interface. When switching the default interface, the codec settings for this interface are recalled.

5.6.3. "AoIP Parameter" page

More details on these parameters in 4.4.1, AoIP Parameter.



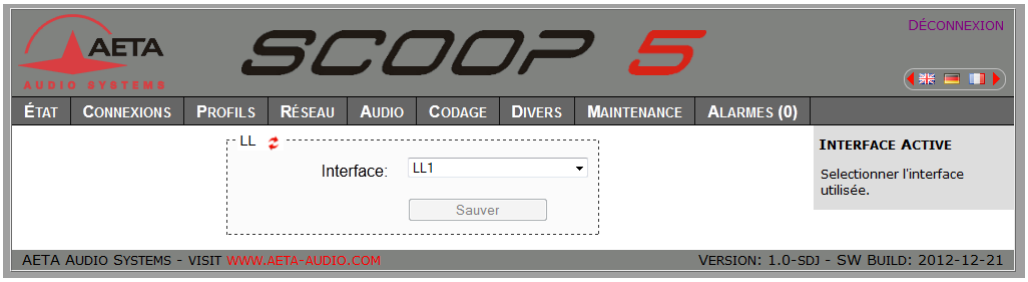
If you select (in the "AoIP Mode" frame) the multicast mode, the page is different:



Remember to **save the changes**, clicking the "Save" button!

5.6.4. LL Parameter page

More details on these parameters in 4.4.1, LL Parameter.



Remember to **save the changes**, clicking the "Save" button!

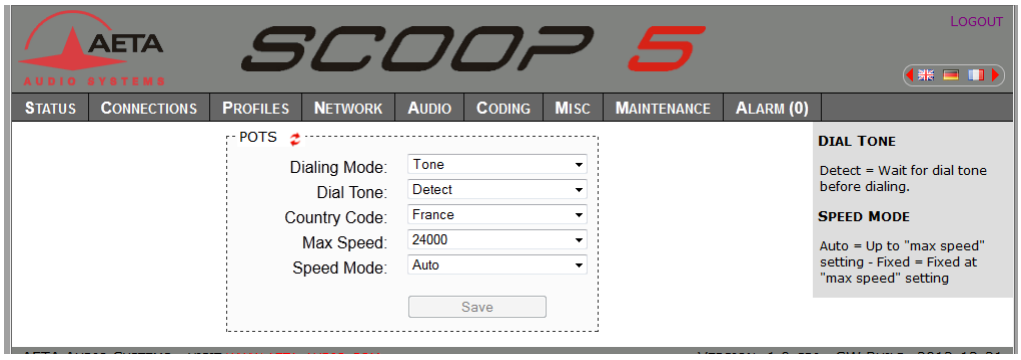
5.6.5. ISDN Parameter page

This page is only visible for units equipped with ISDN interface(s). *See more details on the parameters in 4.4.1, ISDN Parameter.*

Remember to **save the changes**, clicking the "Save" button!

5.6.6. "POTS Parameters" page

This page is only visible for units equipped with the POTS option. *See more details on the parameters in 4.4.1, POTS Parameter.*

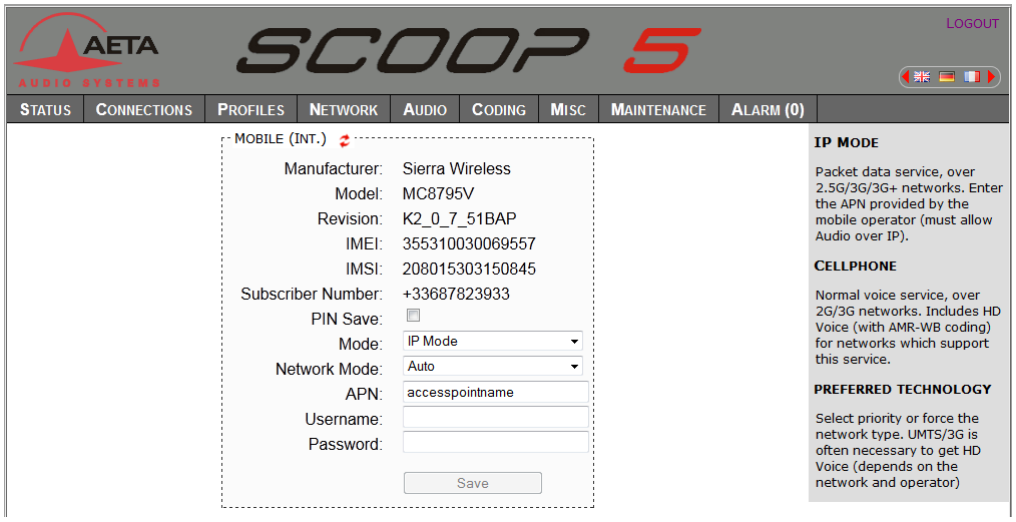


Remember to **save the changes**, clicking the "Save" button!

5.6.7. "Mobile Parameter" page

This page is only visible for units equipped with the "wireless" option. See more details on the parameters in 4.4.1, Mobile Parameter. You cannot access all settings unless the PIN code has been entered (otherwise the view is partial only).

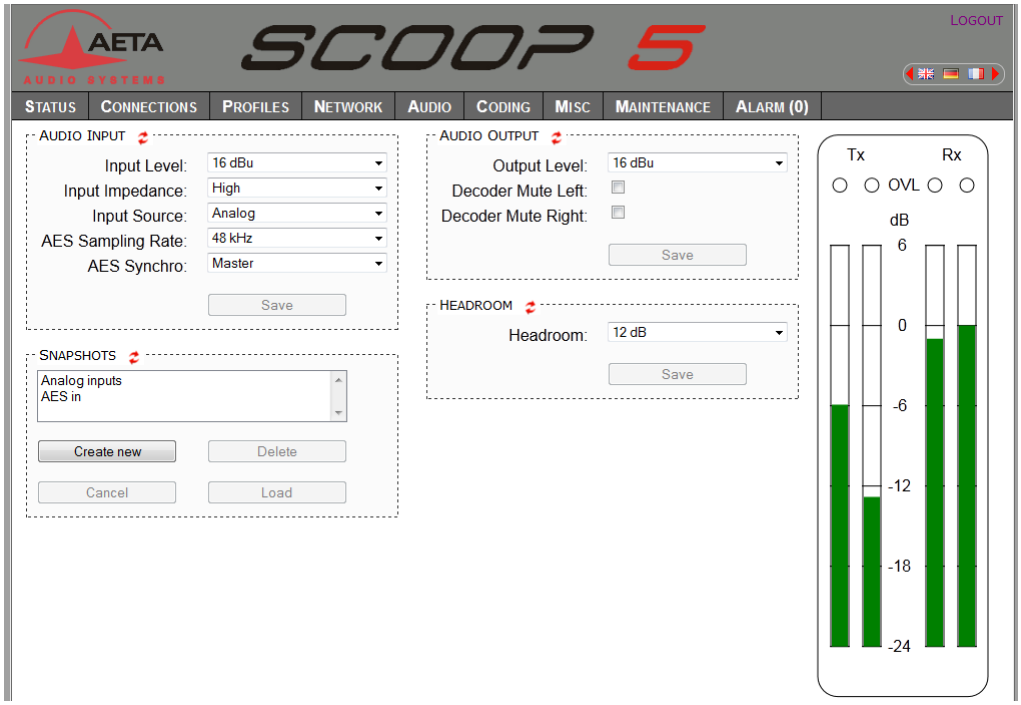
ⓘ *If you have plugged a USB mobile access device, it is also assigned a specific parameter page.*



Remember to **save the changes**, clicking the "Save" button!

5.7. "Audio" tab

See all details for these parameters in 4.4.3, **Audio** sub-menu. Like the "**Status**" page, this page shows the audio level transmitted/received to/from the network.



After changes on the audio parameters, make sure to **save the changes**, clicking the "Save" button.

This page includes the management of *snapshots*, memorizing the parameters controlled in this page.

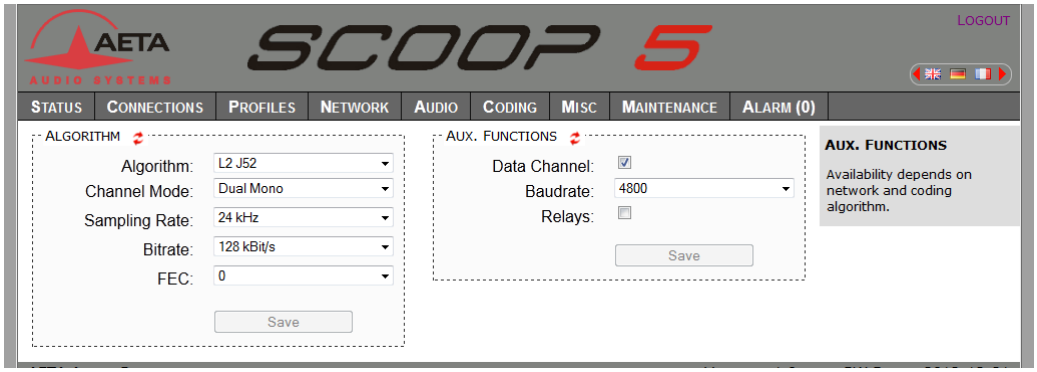
To **create** a snapshot based on the current settings, click the "**Create new**" button, and enter a name for the snapshot.

The other buttons can be used after selecting an existing snapshot. Click a snapshot: its parameters are recalled in the fields of the page, but yet no change is done (the "Save" buttons warn of this). Then you can:

- Click the "**Cancel**" button to simply come back to the previous situation (*also possible using the* ' *icons, but one frame at a time*).
- Click the "**Load**" button and directly apply the snapshot (*also possible using the "Save" buttons, but one frame at a time).*
- Click "**Delete**" to remove the selected snapshot from the list of snapshots.

5.8. "Coding" tab

The "Coding" tab groups the selection of the algorithm and its parameters, as well as the auxiliary functions. See details on all these parameters in 4.4.2, **Algorithm** sub-menu and 4.4.7, **Misc** sub-menu.

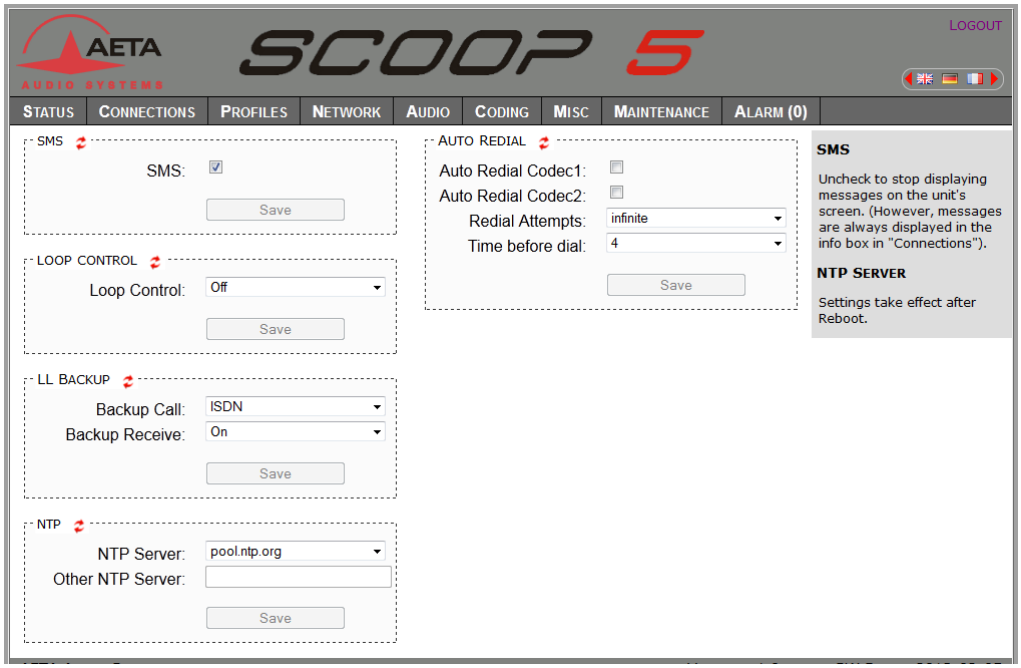


The available coding algorithms depend on the current network interface and the installed options, such as e.g. the AAC codecs. The coding parameters depend on the algorithm and the network interface. This is also true for the auxiliary functions.

Remember to **save the changes**, clicking the "Save" button!

5.9. "Misc" tab

This Misc tab groups several settings that can be found in the Misc sub-menu of the front panel user interface (see 4.4.7, **Misc** sub-menu): auto redialing configuration and activation, loop control activation, SMS reception enabling/disabling…



It is also possible to configure a NTP server: this kind of server provides a time reference (from the Internet or a server on the local network), useful and recommended for the absolute time stamping of the events in the log (see further 5.10.5, "Event log" page). If such a server is available and accessible via the Ethernet interface, enter its address in the "NTP Server" field. A few public servers are also proposed in the dropdown list[1].

ⓘ *A reboot is needed for this setting to become effective!*

ⓘ *Remember to* ***save the changes****, clicking the "Save" button!*

[1] Note: to be able to use these servers 1) access to the Internet from the codec must not be restricted, 2) a DNS must be set so the symbolic names can be resolved.

5.10. "Maintenance" tab

This tab leads to a selection between several pages dedicated to maintenance aspects.

5.10.1. "Login data" page

This page is for setting the password for logging in the html pages. In a classic way, to set up a password you must first enter the current one, and then enter the new one and confirm it.

Remember to **save the change**, clicking the "Save" button!

ⓘ *Reminder: this password has no relationship with that for unlocking the restricted access on the keypad/display front panel interface.*

If you lose the password: using the front panel interface of SCOOP 5, perform a complete reset of the settings, or a complete "factory reset". The passwords (front panel and html access) are then reset as well as the other parameters in the unit.

5.10.2. "Tests" page

This page provides the same choice as the **Maintenance** / **Tests** sub-menu (see in 4.4.8 and 2.5.8, Test functions, the description, of the various test loops available).

ⓘ *Do not forget to disable the test loops (setting "None") to come back to normal operation!*

5.10.3. "System update" page

This page allows to upload a system update file into the unit and update the unit's firmware.

For an update, the file must be provided to you by AETA (or possibly your distributor/dealer). The procedure is rather simple:

- From this page, click "Browse…", find and select the update file.
- Click the "Update" button.
- The process begins, and a warning message is also displayed on the SCOOP 5 front panel.
- Wait for the completion of the update. Normally the unit should reboot by itself at the end of the process.
- Once the unit has restarted, perform a complete "Factory reset".

This procedure requires maximum care, because such update always implies some risk. Please note these additional recommendations:

- Beforehand backup/export the recorded profiles and snapshots, if you would like to use them again afterwards.
- Make sure the link between the control computer and the SCOOP 5 is stable (no undesirable interruption during the process).
- Make sure the SCOOP 5 is not powered off and its Ethernet interface is not disconnected during the process, do not use its keypad during the process.
- Re-import your profiles and snapshots after the update. *But be aware that, although we always try to keep a maximum upward compatibility, AETA cannot guarantee the re-usability and consistency of these memories after a firmware update.*

5.10.4. "Reset" page

This page provides two functions: clearing the settings and/or memories/profiles recorded in the unit, and rebooting the unit.

You can reset the unit completely or only clear a category of data; the available options are:

- "Factory reset": all parameters are brought back to their default setting, and all memories (profiles, snapshots) are deleted.
- "Reset settings": all parameters are brought back to their default setting, but all memories are kept: profiles and snapshots.
- "Reset snapshots": all snapshots are cleared form the memory.
- "Reset profiles": all profiles (remote and local) are cleared form the memory.

The page also allows you to reboot the unit by remote, clicking the "Reboot" button. The device is actually reboot when you confirm by clicking "OK" in the dialog box which then shows up[1]. This is equivalent for SCOOP 5 to an off/on power cycle. Of course, you must wait for the return to the normal operative state before regaining control via the html pages.

[1] If you mistakenly clicked the "Reboot" button, it is still possible to cancel this: close the page without clicking "OK" (do not hit the computer's *Esc* key neither).

5.10.5. "Event log" page

This page displays a history if the system events, which can be useful for operational records, or for investigating operation issues, etc.

Events are logged by the codec in its internal memory card; they are recorded in plain text (unformatted ASCII), and the web page displays the 200 last lines of this log (with a scroll bar).

You can select the detail level for this history, depending on the target application:

- "Coarse": only essential events are logged, such as reboots, link set up, alarms… This type of history can be useful as a link record, and gives a very synthetic overview.
- "Normal": gives more details, and this is the default setting, suitable for most operation needs.
- "Debug": absolutely all events are logged; this makes a very "verbose" and technical history, reserved for testing and fixing issues.

SCOOP 5 can also send event messages to a SYSLOG server is such server is available on the network: enter the address or name of this server and click the "Apply" button. From then on, SCOOP 5 sends the designated server all events, regardless of the detail level selected as described above. The two "histories" run in parallel: on one hand messages to the SYSLOG server, on the other hand the "filtered" events (depending on the detail level selected) recorded in the SCOOP 5 log file.

Events are time-stamped (date, hour, minute, second) from the internal clock of the unit. This clock is not backed (no battery inside the unit), but the unit can synchronize at boot time using the NTP protocol. The address of such server must be set in the "Misc" page (see 5.9 above).

ⓘ *Important notice: time stamps are universal time (UTC), hence they take no account of geographical location, neither any DST (daylight saving time). Remember to add the appropriate time difference when analysing the logs.*

Under the log window you can find some control buttons:

- "Hold": normally the event window scrolls at any time while new events occur. Click this button to stop this and freeze the display. The button becomes "Continue". *The events are still logged, only the page update is interrupted.*
- "Continue": resumes the automatic updating of the event display.
- "Refresh": click this button to update the display and show the most recent events. This button is relevant if the display has been frozen (with "Hold"), in order to refresh the event display on demand.
- "Save logfile as": allows you to download on the control computer the history file. This takes place as a typical download. Depending on the browser and its settings, you can possibly select the file destination, name…

5.10.6. "Setup Transfer" page

This page provides a means to backup all the settings of the unit into a file, and reciprocally to restore a complete configuration from a file previously saved in this way.

Partial exports are also possible, selecting the items you wish to include: remote profiles, local profiles, snapshots, settings…

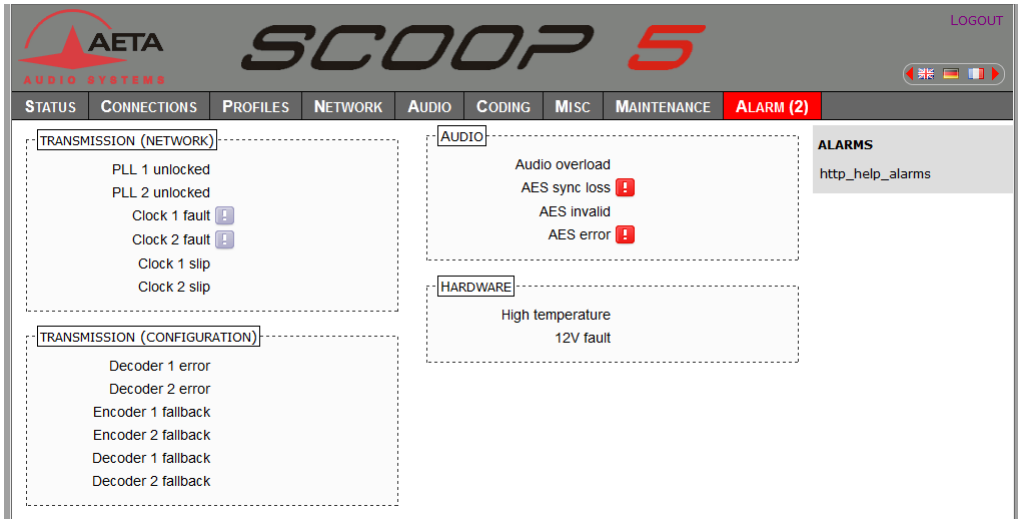
The left part of the page is dedicated to "exporting" the device configuration: the configuration is then saved in a file, downloaded in the control computer. Before doing the export you can select which type of data is to be included in this backup/export: check the desired categories (multiple choice allowed), and click "**Export**". The rest is carried out like a typical download. Depending on the browser and its settings, you can possibly select the file destination, name…

The right side is for "importing" the device configuration: click "**Browse…**" to select the file to be imported, and launch the process by clicking "**Update**". The file must be a file previously exported from a SCOOP 5 (same unit, or another). The settings or memories included in the file are respectively applied to the codec, or recorded in its memory of profiles/snapshots.

Warning: the export files can be imported into a device with the same firmware version, but there is no guarantee of "portability" of the configuration files from one firmware version to another. In other terms, the outcome is not guaranteed when importing into a unit files which were exported from a unit with a different firmware version. We recommend to carefully check the settings in such case, after importing the file.

5.11. "Alarm" tab

The "Alarm" tab switches to red as soon as at least one anomaly is detected. In addition the number of abnormal conditions is shown, like in the example below with two alarm conditions detected:



Clicking the tab, you reach this "Alarm" page showing all the alarm conditions. All possible issues are listed, but only those actually detected are highlighted with the icon.

The issues are grouped in three areas:

- Alarms related to "Transmission": essentially issues related to the decoder synchronization. In addition, for ISDN links, there may be encoder or decoder fallback cases ("fallback" is the situation when the encoder or decoder configuration actually set on the link differs from the one which was initially programmed or expected).
- Alarms related to the audio interfaces: audio clipping (minor alarm), AES input errors.
- Hardware alarms (internal alarms): excess temperature, power failure.

6. Technical characteristics

6.1. Characteristics of interfaces

6.1.1. Analogue audio inputs

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated.

The inputs are balanced type, using 3-pin female XLR sockets.

Maximum input level:	adjustable from 0 to +22 dBu ± 0.3 dB
Nominal input impedance:	600 Ω or 10 kΩ (menu setting: **Setup** / **Audio** / **Analog** / **Input Impedance**)

6.1.2. Analogue audio outputs

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated.
The outputs are balanced type, using 3-pin male XLR sockets.

Maximum output level:	adjustable from 0 to +22 dBu ± 0.3 dB
Nominal load impedance:	600 Ω or 10 kΩ
Output impedance:	<50 Ω
Symmetry:	> 40 dB (ZL = 150 Ω)

6.1.3. Digital audio input and output

These interfaces comply with recommendations:

- AES3-1992
- EBU Tech. 3250-E
- CCIR Rec. 647

They support (in genlock mode) a sampling rate from 28 to 96 kHz. In master mode, the unit can be set in one the following sampling rates: 32, 48 and 96 kHz.

6.1.4. Headphone output (front panel)

This output (6.35 mm jack on front panel) is for the connection of a 32 Ω headphone. It is also possible to plug a high impedance headphone; however, the maximum available power will be lower.

6.1.5. Ethernet Interface

This RJ45 socket has standard Ethernet pinout (for use of a normal “straight” cable to an Ethernet hub or switch). The interface normally operates in 100BaseT full-duplex mode, with auto negotiation, but it is possible to force other configurations.

The installation and operation of this function is detailed in 3.4, Initial setup of the Ethernet interface.

6.1.6. Main X24/X21/V11/V35 interface

The X24/V11 interface uses a 15-pin male Sub-D connector. The following table shows the pinout.

Signal		Pin		Signal	
Frame ground		1			
Transmitted data	Ta	2	9	Tb	Transmitted data
		3	10		
Received data	Ra	4	11	Rb	Received data
Indication	Ia	5	12	Ib	Indication
Received clock	Sa	6	13	Sb	Received clock
		7	14		
Electrical ground		8	15		

The codec does not transmit a C signal, and the I signal has no effect.

The codec can also be connected to a V35 interface; a specific adaptation cable is needed in such case. The connection is described in Annex (7.4, V35 interface adaptation).

6.1.7. "Alarm + X24/X21" interface

This interface uses a 15-pin male Sub-D connector. The following table shows the pinout.

Signal		Pin		Signal	
Internal alarm - Common	**IA-Com**	1			
Transmitted data	Ta	2	9	Tb	Transmitted data
Internal alarm - NC	**IA-C**	3	10	**IA-O**	**Internal alarm - NO**
Received data	Ra	4	11	Rb	Received data
Indication	Ia	5	12	Ib	Indication
Received clock	Sa	6	13	Sb	Received clock
External alarm - NO	**EA-O**	7	14	**EA-C**	**External alarm - NC**
Electrical ground		8	15	**EA-Com**	**External alarm - Common**

The bold text refers to the alarm contacts. Both are form-C type. The "NO" terminal is open when the alarm is set, otherwise it is connected to the "Common" terminal. The "NC" terminal is connected to the "Common" terminal when the alarm is set, otherwise it is open.

The current and voltage handling capabilities of the relays (static type) are:

Maximum output current: 120 mA

Maximum output voltage: 350 V peak

Resistance of output loop: $< 35\ \Omega$

Isolation $> 1500\ V_{RMS}$

The codec does not transmit a C signal and the I signal has no effect.

The codec can also be connected to a V35 interface; a specific adaptation cable is needed in such case. The connection is described in Annex (7.4, V35 interface adaptation).

6.1.8. ISDN interfaces

These interfaces are not available for all versions.

The sockets are ISDN S0 interfaces (BRI), the supported protocols are ETSI (Euro RNIS), NI-2 or NTT.

Socket #1 has a slightly different pinout, as contacts 1 and 2 (unused on a standard ISDN socket) are allocated to the POTS interface.

6.1.9. POTS/PSTN interface

Interface available as an option.

The telephone interface (2 wires) is combined with the ISDN socket #1, using contacts normally unused of the S0 socket. The pinout on this socket is as follows:

Line	Pin	Signal
POTS	1	POTS line
	2	
ISDN	3	T+ (transmission)
	4	R+ (reception)
	5	R- (reception)
	6	T- (transmission)
	7	-
	8	-

An **adapter cable** is provided with the POTS option, in order to ease the installation. This adapter features on one end an RJ45 plug, to be inserted in the ISDN socket #1 of the SCOOP 5. On the other end it features a small block with two sockets:

- An RJ45 socket, to be connected to an S0 line (BRI) using a « straight » RJ45 cable;
- An RJ11 socket, to be connected to the POTS telephone line.

6.1.10. Antenna sockets (mobile networks)

Available on units equipped with the "wireless" option. See also chapter 2.1.5: Mobile network access.

These sockets are used to connect one or two antennas (one multiband antenna is included with the mobile network option).

The interfaces are male SMA (female contact), 50 Ω impedance.

The internal mobile access module can operate in following bands:

Frequency band	850 MHz	900 MHz	1800 MHz	1900 MHz	2100 MHz
2G GSM/EDGE					
3G/3G+ UMTS/HSDPA/HSUPA/HSPA/HSPA+					

Specific versions exist for access to 4G/LTE networks (contact AETA).

6.1.11. Serial remote control interface

This interface uses a 9-pin female Sub-D connector on the rear panel. This is a V24/RS-232 type interface with only Tx and Rx signals (no flow control). The following table indicates its pinout (DCE type pinout).

Pin		Function	
2	Rx	V24 data to the PC	Output
3	Tx	V24 control data, from the PC	Input
5		Ground	
Other		Not connected	

The interface is configured as follows: 4800 bauds, 8 bits, no parity, one stop bit.

6.1.12. Serial data interface (“Data”)

This V24 interface uses a 9-pin female Sub-D connector on the rear panel. Like for the remote control interface, only Tx and Rx are used, there is no flow control, and the pinout is of DCE type.

Broche		Fonction	
2	Rx	Received V24 data	Output
3	Tx	Transmitted V24 data	Input
5		Ground	
Other		Not connected	

The data interface is configured as follows: 8 bits, no parity, one stop bit, no flow control. It is possible (see menu **Tools** / **Misc** / **Aux. Functions**) to activate the interface and to configure its baud rate (300 to 9600 bauds). However, the maximum allowed baud rate depends on the audio coding used (see 2.4, Auxiliary functions).

6.1.13. Relay transmission interface ("AUX" socket)

The relay transmission interface (refer to 2.4.1, Transmission of isolated relays) is available on the 25 pin female sub-D "AUX" Socket. It includes two isolated current loop inputs and two dry contact outputs.

The following table shows the pinout of the socket for this function:

Pin	Function
13	Output loop n°2 (b)
25	Output loop n°2 (a)
12	Output loop n°1 (b)
24	Output loop n°1 (a)
11	Input loop n°1 (b)
23	Input loop n°1 (a)
10	Input loop n°2 (b)
22	Input loop n°2 (a)
9	+5V of internally supplied power supply
21	0V of power supply

All loops are isolated and bi-directional (free polarity).

The characteristics of the input loops are:

Input loop control current:	6 mA	(max. 100 mA)
Resistance of input loop:	~ 560 Ω	(current limiting series resistor)
Input loop isolation:	> 1500 V_{RMS}	

A +5V to +12V source may be connected directly on an input loop, because the internal series resistor is dimensioned for this purpose. For a higher voltage source, it may be necessary to limit the input current.

The characteristics of the output loops are:

Maximum switching voltage:	350 V peak
Maximum switching current:	120 mA
Resistance of output loop:	< 35 Ω
Output loop isolation:	> 1500 V_{RMS}

The 5V power supply is available from the unit to power a low-consumption device (maximum 300 mA current consumption), e.g. to power the input loops or LED indicators connected to the output loops.

6.1.14. Loop control interface ("AUX" socket)

The 25 pin female sub-D "Aux." socket includes isolated current loop inputs and dry contact outputs, that can be used to remotely control the calls and indicate the link status:

- The input loops have an effect only if the "loop control" function is enabled (see 3.5.3, Loop control). The output loops are always operative.
- Activating the input loop #1 triggers an ISDN or IP call on the codec (codec 1 only if the unit is configured as a double ISDN codec); de-activating the loop releases the line.
- Activating the input loop #2 triggers an ISDN call on codec 2 if the unit is configured as a double ISDN codec; de-activating the loop releases the line. This loop has no action in single codec mode.
- Output loop #1 is closed while an ISDN or IP connection is running, or while codec 1 is linked if the unit is configured as a double ISDN codec;
- Output loop #2 is closed while an ISDN connection is running on codec 2, if the unit is configured as a double ISDN codec;

The following table shows the wiring of the socket for this function:

Pin	Function
17	Input loop n°2 (a)
5	Input loop n°2 (b)
18	Input loop n°1 (a)
6	Input loop n°1 (b)
19	Output loop n°2 (a)
7	Output loop n°2 (b)
20	Output loop n°1 (a)
8	Output loop n°1 (b)
21	0V of power supply
9	+5V of internally supplied power supply

All loops are isolated and bi-directional (free polarity).

The characteristics of the input loops are:

Input loop control current:	6 mA	(max. 100 mA)
Resistance of input loop:	~ 560 Ω	(current limiting series resistor)
Input loop isolation:	> 1500 V_{RMS}	

A +5V to +12V source may be connected directly on an input loop, because the internal series resistor is dimensioned for this purpose. For a higher voltage source, it may be necessary to limit the input current.

The characteristics of the output loops are:

Maximum switching voltage: 350 V peak

Maximum switching current: 120 mA

Resistance of output loop: $< 35\ \Omega$

Output loop isolation: $> 1500\ V_{RMS}$

The 5V power supply is available from the unit to power a low-consumption device (maximum 300 mA current consumption), e.g. to power the input loops or LED indicators connected to the output loops.

6.1.15. Coordination channel interface ("AUX" socket)

In addition to the loop control and relay transmission interfaces, the (optional) coordination channel input and output are available on the 25-pin female sub-D connector ("AUX" Socket on the rear panel), with pinout as indicated hereunder.

The input and output are balanced floating signals, transformer isolated.

Maximum level: 9 dBm

Impedance: 600 Ω

Nominal bandwidth: 300 – 3400 Hz

Pin	Function
1	Coordination channel output (-)
14	Coordination channel output (+)
2	Frame ground
15	Coordination channel input (+)
3	Coordination channel input (-)
16	Frame ground

6.1.16. "Digital I/O" interface

This 15 pin female sub-D socket provides the GPIO signals described in 2.4.2, Transmission of GPIO. The pinout is shown in the following table:

Pin	Function
1	Signal and power ground
9	
2	GPI 3
10	GPO 3
3	GPI 4
11	GPO 4
4	GPI 5
12	GPO 5
5	GPI 6
13	GPO 6
6	GPI 7
14	GPO 7
7	GPI 8
15	GPO 8
8	+5V of internally supplied power supply

Note: the GPIO index starts at 3 because indexes 1 and 2 are allocated to the relays (see above in 0).

The GPIO signals do not have galvanic isolation as the relays. They feature the following characteristics:

Characteristics	Min	Nominal	Max	Notes
GPI: voltage (low level, active)	-0.5 V	0 V	1 V	
GPI: current (@ 0 V)			110 µA	
GPI: voltage (high level, inactive)	3 V	5 V	7 V	[1]
GPO: voltage (low level, active)	0 V	0 V	0.55 V	
GPO: current (low level, active)			32 mA	
GPO: voltage (high level, inactive)		5 V	6.5 V	[2]

[1] An input pull-up ensures a high level if the GPI are "open" (not connected).

[2] The GPO are "open drain", and must be "pulled up" (to 5 V nominal) for the high level, but an internal pull-up ensures a 5 v voltage when they are inactive, even with no external polarization.

6.1.17. AC power supply

The unit is connected to mains via an IEC 3 pin socket, and accepts a 85-263Vac, 47-63 Hz AC source. Protection is provided by a resettable fuse.

6.1.18. DC power supply (option)

When the 12V DC option is installed, the unit features, in addition to the mains socket, a 4 pin male XLR socket for connecting a 12 V DC supply (non isolated). The unit includes overvoltage and overcurrent protection (resettable fuse on the DC input).

The DC supply is redundant with mains power, with priority to mains whenever it is present [1].

Characteristics	Min	Nominal	Max	Notes
Supply voltage	10 V	12 V	17 V	[1]
Supply current	0.5 A		2 A	[2]

[1]: Beyond 15 V the DC supply takes precedence, and provides power to the unit even if AC power is available.

[2]: The power consumption may vary in a wide range depending on the input voltage, the installed options, and the operating mode.

Connections on the XLR socket: Ground is connected on pin 1, and pin 4 must provide the +12 V voltage.

6.2. Audio performance

The audio performance in this part applies to the system without coding/decoding, and excluding the coordination channel. The additional effect of the audio encoding and decoding on audio performance depends on the coding algorithm used and its parameters.

Except when differently stated, the following measurements are done at a +6 dBu input level and on the AD/DA path, with maximum input and output level set at +16 dBu.

6.2.1. Transmission gain

The drift in time of the gain from the input to the output of the codec is less than ± 0.3 dB.

6.2.2. Amplitude-frequency response

All measurements are done with a +6 dBu input signal, and a reference frequency of 1020 Hz. The measurements are done with a loopback before coding/decoding, so the possible effect of compression has no influence.

For a 48 kHz sampling rate:

Frequency range		Tolerance	
0 Hz	20 Hz	-∞	0 dB
20 Hz	40 Hz	-0.2 dB	0.1 dB
40 Hz	20 000 Hz	-0.1 dB	0.1 dB

For a 32 kHz sampling rate:

Frequency range		Tolerance	
0 Hz	20 Hz	-∞	0 dB
20 Hz	40 Hz	-0.2 dB	0.1 dB
40 Hz	15 000 Hz	-0.1 dB	0.1 dB

6.2.3. Group delay distortion

Taking the minimum group delay as reference, the group delay distortion on the AD/DA path is always less than 1 ms.

6.2.4. Idle channel noise

Background noise is measured with no audio modulation (idle channel), with maximum input and output level set at +16 dBu, through the whole encoder-decoder chain (wide band coding, with 48 or 32 kHz coding frequency).

Maximum noise level[1]: - 55 dBm (quasi-peak detection, CCIR weighting)
(or - 61 dBq0ps)

This result in a signal to noise ratio (SNR) of more than 71 dB.

When the maximum input and output level is set at another level, both the signal and noise levels are shifted but the SNR remains in the same range.

[1] Worst case for all types of algorithms; MPEG algorithms performs better than the others

6.2.5. Total distortion vs frequency and level

Total distortion relative to maximum level (or THD + N) is less than –82 dB over the whole audio bandwidth (20 – 20 000 Hz). This performance holds for audio signals from –80 dB to –1 dB relative to the maximum level (+16 dBu).

6.2.6. Crosstalk

Crosstalk is less than -80 dB over the whole bandwidth.

6.2.7. Gain and phase difference between channels

The gain difference between channels is less than ± 0.3 dB over the whole bandwidth, for any sampling frequency.

The phase difference between channels is less than ± 3 degrees over the whole bandwidth, for any sampling frequency.

6.3. Network protocols and ports

The SCOOP 5 implements or complies with the following protocols (non exhaustive list):

- Physical and link layers: Ethernet, 100BaseT, 10BaseT
- Network/Transport layers (IPv4): TCP/IP, UDP/IP, RTP/IP
- Application: HTTP, Telnet, DHCP, STUN, NTP, SYSLOG
- Audio transmission: SIP signalling, SDP, RTP, RTCP, RFC3550/3551, RFC3640
- Compliant with EBU recommendation Tech 3326 (interoperability of audio codecs for contribution)

The ports used by the device are the following:

Type	Port	Designation	Dir	Notes
TCP	80	HTTP	⇆	Embedded html server
	6000	Control	⇆	Remote control ("command line" mode); used by Scoop4Man and TeleScoop
UDP	123	NTP	⇆	For querying an NTP server
	514	SYSLOG	→	For sending messages to a SYSLOG server
	3478	STUN	⇆	For querying a STUN server
	5004	RTP	⇆	For audio streaming
	5005	RTCP	⇆	For audio streaming
	5060	SIP	⇆	SIP signalling
	6000	Multicast/description	⇆	Multicast stream description channel
	6001	Multicast/audio	⇆	Multicast stream transmission channel

*Ports shown in **bold** can be changed from the default values shown in the table.*

6.4. Power supply

The codec operates from mains 85-263Vac, 47-63 Hz.

As an option the product can also be powered from 12V DC source.

The maximum power consumption is 20 W (depending on version and installed options).

6.5. Dimensions and weight

The unit is a 19 inches frame of 1U height (44 mm or 1.75") and 265 mm overall depth (12.5").

Its maximum weight is about 3.8 kg (depends on version and installed options).

6.6. Environmental characteristics

The equipment operates over a 0°C to 45°C ambient temperature range (32°F to 113°F), and a 5% to 90% humidity ratio range.

The SCOOP 5 complies with "CE" directives regarding safety and EMC.

- Safety: compliance with EN60950
- EMI: radiated emissions complying with EN55103-1
- Susceptibility: compliance with EN55103-2

6.7. Options

The basic version of SCOOP 5 includes digital leased line interfaces (X24/V11/V35) and an Ethernet interface for IP transmission. Many available options can be added to complement the capabilities of the SCOOP 5.

Note: for IP or leased line transmission, and when the application does not require a front panel interface, AETA also proposes the SCOOP 5 IP, which features a front panel with no control element and is controlled only by remote. For more information refer to AETA web site and the dedicated documentation for this product.

6.7.1. Network options

For adding more types of network interfaces, the following options are available:

Ordering code	Option	Description
80 00 211 01	2B ISDN	Addition of one ISDN interface, for 64 or 128 kbit/s transmission (1 or 2 B channels)
80 00 211 02	4B ISDN	Addition of two ISDN interfaces, for 64 to 256 kbit/s transmission (1 to 4 B channels)
80 00 193 21	Wireless	Addition of the mobile network access
80 00 193 01	POTS codec	Addition of the POTS/PSTN interface
80 00 193 31	Wireless + POTS codec	Mobile and POTS interfaces

6.7.2. Other options

Ordering code	Option	Description
80 00 192 51	AAC	Addition of AAC coding algorithms: AAC-LC, HE-AAC and HE-AAC v2
80 00 192 01	TDAC	Addition of the TDAC codec
80 00 194 01	12V DC power	Additional 12V DC power supply input
80 00 196 01	Coordination channel	Auxiliary coordination channel

6.8. Accessories and related products

The SCOOP 5is delivered with a mains cord and a CAT5 Ethernet cable.

The POTS option includes a cable adapter which splits the ISDN socket #1 into an RJ45 socket (ISDN) and an RJ11 socket (POTS).

The "wireless" option is delivered with a multiband antenna, with a 3 m cable and an SMA plug.

An adapter is available if needed for using a µSIM card instead of a standard SIM card.

Along with the coordination channel option, a specific cable is delivered, which provides XLR plugs for the coordination channel input and output (input on a female plug, output on a male plug).

For remote controlling SCOOP 5 units from a PC, the TeleScoop configuration and supervision software is available (for a Windows computer). The TeleScoop software also allows to control the other rackmount codecs of the Hifiscoop, Scoop 3, Hifiscoop 3 and Scoop 4+ ranges.

SCOOP 5 can also be controlled by several third-party software tools such as Codec Live, MDC.Net...

For an additional Ethernet interface, usable for remote control from a separate sub-network, it is possible to plug on the USB socket a USB/Ethernet adapter, such as the LogiLink UA0144.

Other compatible devices may be available later, in which case you can find them on our web pages; consult our site www.aeta-audio.com .

7. Annexes

7.1. Additional information on the algorithms and protocols used

7.1.1. Auxiliary data in the MPEG frames

The auxiliary data are used for the following purposes:

- Reed-Solomon error detection and correction (J52 standard)
- Data channel
- Other auxiliary information: relay transmission, and/or auxiliary audio channel. The insertion of this auxiliary information is an extension (AETA proprietary format) to MPEG. However, the frame structure remains compliant.

7.1.2. Reed-Solomon encoding

In order to cope with possible transmission errors in the network, Reed-Solomon error correction coding can be added, compliant with J52 recommendation. Four correction modes are available in the SCOOP 5:

- Mode 0: no error correction, Reed-Solomon coding disabled
- Mode 1: protection of only the control information and scale factors in the MPEG frame, low redundancy
- Mode 2: protection of the whole frame, moderate (2.5 %) redundancy
- Mode 3: protection of the whole frame, high (10 %) redundancy

Higher redundancy increases the protection against errors, but slightly degrades the audio quality, as redundancy takes up part of the bit rate that could be allocated to audio coding.

Most often, for a normal quality transmission link, mode 1 is sufficient and it consumes little bit rate from the compressed data, so it hardly impacts the audio quality. Although J52 does not apply to leased line connections, this error protection technique is also implemented in the SCOOP 5 in leased line mode.

7.1.3. H221 framing

H221 defines a framing structure that allows byte synchronisation recovery in leased lines, and the transmission of control data along with the main data.

7.2. Overview of the SIP protocol

7.2.1. What is SIP?

SIP is for Session Initiation Protocol, a protocol specified by the IETF for establishing media transmission sessions. SIP is considered the communication protocol of the future by most vendors, and as such it has deep influence on the VoIP applications.

As a signalling protocol, SIP brings methods and techniques to solve the issues related to the establishing of an audio link. Almost as important, it is a recognised standard, implemented on many network devices and systems. Using SIP helps you build modular and really evolutive systems, not being tied to a single vendor.

The SIP protocol is an essential requirement in the EBU Tech 3326 recommendation (a.k.a. "N/ACIP" recommendation, from the name of the EBU workgroup who elaborated it).

7.2.2. Setting a link with SIP

Let us look at an example (diagram below): a reporter on the move with a Scoopy+[1] wants to make a call to a SIP compliant codec located in the main station. The reporter may be at home, or at another location, not necessarily known in advance.

Once the Scoopy+ is on and connected to the network, it will register itself ❶ to a SIP "registrar". This registrar can be located on the LAN of the radio house, but it may as well be elsewhere in the network. Then the registrar "knows" where the Scoopy+ is, what its IP address is. On the radio house side, a similar process takes place ❷.

To call the codec of the radio house (e.g. a SCOOP 5), the reporter just needs to know its SIP address, which can be like studio12cod@radiomcr.com (indeed very similar to an e-mail address). To call the unit, the reporter has to select the preferred audio coding mode on the Scoopy+ (e.g. mono G722), then call the remote unit, simply using this SIP address (SIP URI).

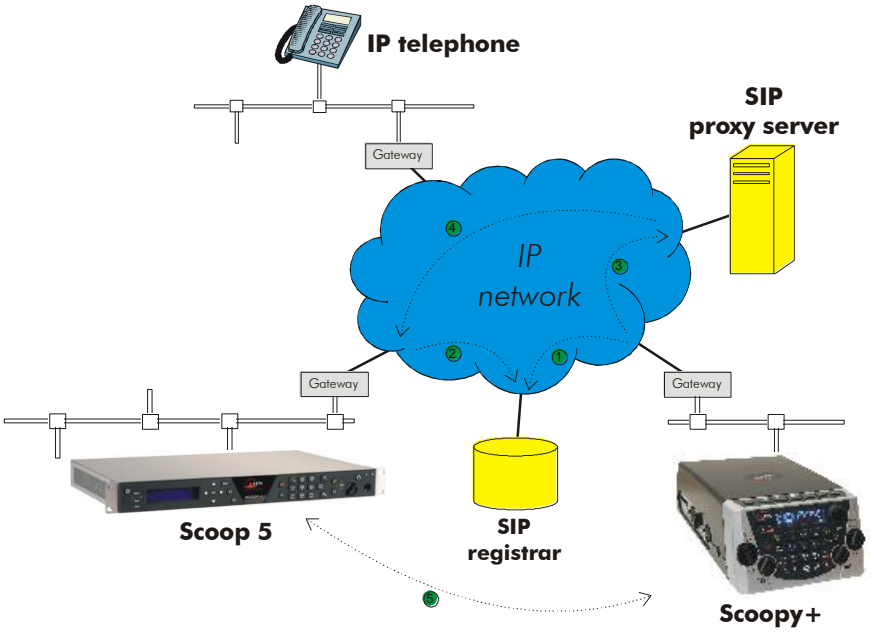


Figure 7 – Setting up an SIP session with server(s)

[1] Scoopy+ is a portable audio codec from AETA; the description here applies to both Scoopy+ and the Scoop 5, as they are both SIP compliant and mutually compatible

What happens then on the network: the Scoopy+ sends the request ❸ (INVITE in SIP protocol) to a proxy server (often the same device is also the registrar). To make things simple, this proxy then relays and routes the request ❹ to its destination. Resolving the SIP URI to a physical network and address uses mechanisms similar to those used for resolving URLs. Several proxys may be involved in cascade to finally reach the desired destination, but this does not have to be known or dealt with by the end devices. The following is like the initiation of a phone call: the IP codec "rings"; if it accepts the call, this is notified to the Scoopy+.

At this stage, the proxy (s) provide the Scoopy+ and the IP codec with all the address data they need for the link, then the actual audio streams can be exchanged ❺ between both units. As a very important feature, the end devices now can exchange data directly; the proxies do not have to be on the path, they are only involved in the setting (and later the ending!) of the session. The codecs will automatically exchange their coding capabilities, and agree on a coding mode, with no further user intervention.

Alternatively, the call can be done from the station to the reporter, in a way very similar to the above. In contrast with ISDN links, the operators at the station do not even need to know where the reporter is located! This is because the registrar deals with this issue.

Note that it is also possible to set a link with a SIP-compliant VoIP phone instead of another codec. This is one of the numerous advantages of using a standard.

7.2.3. Setting a link without a SIP server

Contrary to a commonplace but wrong idea, the SIP protocol does not impose using a SIP server. The SIP *protocol* can be used without a SIP *server*, i.e. it is possible to set "peer to peer" links without involving one or more SIP *registrars* or *proxies*. In such a case the process is more direct, and the codec contacts the remote destination with no intermediate entity. However, there are some drawbacks with such type of session:

- Without a *registrar*, the identification of the destination is its IP address; this is not as "stable" as a SIP URI, as it may change depending on the location, or time (with dynamic addressing).
- The two codecs must have open access to the Internet, or otherwise the link must be made possible by unlocking if necessary the appropriate routes and ports (conversely, when using a proxy, such allowances can be restricted to the server connection, for a better control on the access security).
- Most often the gateways perform address translation (NAT), which is an *a priori* issue for UDP protocols such as implemented for real time audio links. Proxies can help dealing with this issue, but without such servers it may be rather difficult to work around such obstacle. *The following chapter provides some recommendations about this.*

7.3. Some methods to deal with NAT routers and firewalls

Problems arise when the desired connection has to go through a NAT router, and/or a firewall, that blocks a direct IP communication.

This is a very common issue, especially if one needs to set up a transfer via the Internet. It is impossible here to describe in details the possible ways to deal with this problem, but the following just shortly discusses some typical solutions. Most probably, a network administrator should be consulted for support, and for granting adequate network authorisations and/or privileges.

The most classical issues are related to:

- The presence of a NAT router on the network path between the codecs.
- The presence of a firewall on this path.

It is always important to have available the information regarding the network organization and to be allowed to access the devices which need to be configured. Hence we highly recommend to involve the persons empowered for such tasks.

7.3.1. Links via a private network

No special problem should be met within a LAN. The operation is also possible with codecs from other manufacturers, provided that they comply with the Tech3326 EBU recommendation (also known as "N/ACIP" recommendation). However you should check for specific settings or preparation possibly needed on such devices.

A wide area network covers a wider geographic range, and the network topology most probably includes routers on the path between the codecs to be linked. However, usually there is not much difference with a local area network.

Note: using a VPN leads to just the same case; the operation is identical as far as the codecs are concerned.

7.3.2. Links through a public network (Internet)

If each of the units has got a "direct" access to Internet with a public address, we are in the same situation as the previous one, functionally speaking (private WAN). The addressing scheme is normally static, as DHCP can rarely be used on a public access. In fact, this situation is very seldom met in the field!

First, the Internet access is usually protected by a firewall which will, as a principle, block a priori the desired connection. In such case exceptions (to the firewall security rules) must be created, that will allow this connection; this has to be done by the person in charge of the network management.

Most often, on one access if not both, the codec accesses the Internet via a NAT router. This router shares Internet access, with one or a few public addresses, among the equipment on the LAN. On this LAN the devices get local private addresses, and the router carries out an IP address translation. Note that:

- As an example, a consumer ADSL modem-router is almost always a NAT router, sharing a single public IP address between the devices connected to the router.
- It is just the same on a 3G/3G+ mobile IP access; the terminals (phones or computers) access the Internet via NAT routing.
- NAT routing is often included in the firewall features; in fact NAT routing somewhat participates to the protection against direct attacks from the outside.

NAT routing is an obstacle to transmission with UDP, mainly for two reasons:

- It does not allow unsolicited data to come in from the outside network. In other words, data input is accepted on a port as an answer to a request from the local network, but an external agent cannot directly initiate the transmission of a packet.
- The terminal units on the LAN only know their private local address. On the other hand, agents implementing the SIP protocol have to communicate to each other the addresses and ports to be used for the media exchanges. Because of the NAT routing, agents do not get the real public addresses, which leads to failure of the session setup attempts.

We are now looking at various methods used to overcome these obstacles.

NAT and use of a STUN server

The STUN protocol is a method which is often successful[1] in helping the agents to discover their public address even when they are "hidden" behind a NAT router. Here is the operation principle:

- A STUN server is used, which is accessible over the Internet;
- The address of this server is programmed into the agent (i.e. the audio codec in our topic);
- The agent queries the server and discovers its public IP address and port number, as seen from outside of the NAT router and LAN;
- This addressing information is then used by the agent for negotiating and setting up a media session.

The STUN server address is programmable in the menu or the html pages of a SCOOP 5 or Scoopy+. Besides, there is also in the menu (keypad and display on the front of the unit) an enable/disable (on/off) selection, without having to clear the server address.

There are many public STUN servers available on the Internet; here are a few examples, valid at the time of writing:

Domain name	Numeric address
stun.aeta-audio.com	85.214.134.163
stun.ekiga.net	77.72.174.163
stun.sipgate.net	217.10.68.152

Examples of STUN servers

It is recommended to check that the server is operative. Moreover, numeric addresses may change, even if the domain name stays the same. A list of servers can also be found on the support page of our web site http://www.aeta-audio.com.

[1] Although not with so called "symmetric" NAT routers

Standard NAT router

Situation: codec A behind a NAT router with no specific programming *(a codec accessing Internet via a mobile network is almost always in such situation)*.

Besides, we assume that the other codec (B) is accessible with a public address.

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B cannot call codec A

Advantages	Drawbacks
Configuration is relatively simple	B cannot call A
No change is needed on the router	
Several codecs can be set behind the NAT router	
Method suitable for mobile network access[1]	

NAT router with DMZ

Situation: codec A behind a NAT router and placed in « DMZ ».

We also assume that the other codec (B) is accessible with a public address.

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B can call codec A, using the public address of the NAT router

Advantages	Drawbacks
Each codec can set up a session	Need to configure the router
A is nearly equivalent to a codec with a direct public access	Only one codec can be set up in this way on a LAN
	A is exposed to external attacks
	The DMZ may be already reserved for other network equipment
	Method not possible for a mobile network access

NAT router with port forwarding

Situation: codec A behind a NAT router and configuration of the router to forward to A the necessary ports.

We also assume that the other codec (B) is accessible with a public address.

Port forwarding to be set on the router[2]:

- UDP 5060 (= SIP port)
- UDP 5004 (RTP port) and 5005 (RTCP port)

[1] Except with symmetric NAT, which is often met for mobile network access

[2] If needed you can change these numbers on the SCOOP 5

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B can call codec A, using the public address of the NAT router

Advantages	Drawbacks
Each codec can set up a session	Need to configure the router
A is nearly equivalent to a codec with a direct public access	Only one codec can be set up in this way on a LAN
	Method not possible for a mobile network access

Use of a SIP server

In addition to the numerous features it brings along, using a SIP proxy server is a very powerful method to solve the issues related to NAT routers, because most SIP proxies are capable to detect the presence of NAT routers and/or deal appropriately with their traversal.

If a SIP server is available, and once the codecs are registered on this server:

- Any registered codec can call another registered[1] codec, regardless whether there is or not a NAT router on the path.
- The identifier (SIP URI) is stable and does not depend on the location of the called agent ("mobility" feature).

It is possible either to use a public server on the Internet, or to install a private server accessible via the Internet.

Advantages	Drawbacks
Each codec can initiate a session Each codec can receive calls	Installation may not be easy (private server)
Identification is simple and location/time-wise stable	Reliability of server questionable (public server)
Security: a private proxy can be linked with a firewall	
Also works with symmetric NAT routers	
Interoperation with telephony over IP	
Method suitable for mobile network access	

For a fast implementation, you can use SIP accounts on AETA's SIP server sip.aeta-audio.com. This server is dedicated to a professional broadcast usage, and housed in a safe site, available 24/7. Contact AETA for subscribing SIP accounts.

[1] Depending on the access control policy, a server may accept « outgoing » calls to third party domains, or accept « incoming » calls from non registered agents.

7.3.3. Summary and reminder of essential rules

The table below sums up the situations where a link can be set up (not using a SIP proxy server) and reminds the needed specific settings:

	Codec A access	Possible calls	Codec B access	Notes
1	LAN	⇒ ⇐	LAN (same)	
2	Private WAN	⇒ ⇐	Private WAN	
3	Internet direct	⇒ ⇐	Internet direct	
4	NAT	⇒	Internet direct	STUN needed for A
5	NAT + DMZ	⇒ ⇐	Internet direct	STUN needed for A
6	NAT + port forwarding	⇒ ⇐	Internet direct	STUN needed for A UDP ports 5004, 5005, 5060
7	NAT	⇒	NAT + DMZ	STUN needed for A and B
8	NAT + DMZ	⇒ ⇐	NAT + DMZ	STUN needed for A and B
9	NAT + port forwarding	⇒ ⇐	NAT + DMZ	STUN needed for A and B UDP ports 5004, 5005, 5060
10	NAT	⇒	NAT + forwarding	STUN needed for A and B
11	NAT + DMZ	⇒ ⇐	NAT + forwarding	STUN needed for A and B
12	NAT + port forwarding	⇒ ⇐	NAT + forwarding	STUN needed for A and B UDP ports 5004, 5005, 5060

ⓘ *Basic rule: Codec behind a NAT router => use a STUN server.*

This allows the codec to set up *outgoing* calls. This is not sufficient to be accessible to connection requests *from* the outside.

ⓘ *Mobile network access without SIP server or VPN => use a STUN server*

ⓘ *NAT + DMZ or NAT + forwarding => incoming calls are possible.*

Incoming calls are not possible behind a NAT router without either such change or a SIP proxy.

ⓘ *SIP server => maximum versatility, at the expense of some initial effort (for installation)*

ⓘ *Reminder: the SIP protocol (always used by AETA codecs) does not impose the use of a SIP server. Codecs can set up point-to-point links using this protocol in the above described conditions. When no SIP registrar is involved, the identifiers are simply the IP addresses of the codecs.*

7.4. V35 interface adaptation

This annex indicates the proper connections to use when interfacing the SCOOP 5 codec to DCE equipment using a V35 interface.

In the connection table below, the three leftmost columns show the pin allocation on the 15-pin connector of the codec.

The two columns on the right indicate the pinout on a 34-pin V35 connector or a 37-pin sub-D connector. Consult the DCE documentation for other connector types.

Only the bold indicated signals need be connected; leave other pins unconnected. However, the frame ground (pin 1) may be used for connecting the shield of the connection cord.

X24 Signals	Pin	Signal direction	Function	V35 signal		Pin number 34-pin connector	Pin number 37-pin connector
G	**8**		**Signal ground**	**102**	**SG**	**B**	**19**
	15						
	7						
	14						
Sa	**6**	←	**Data clock**	**115a**	**RETA**	**V**	**8**
Sb	**13**	←	**Data clock**	**115b**	**RETB**	**X**	**26**
Ia	5	←					
Ib	12	←					
Ra	**4**	←	**Received data**	**104a**	**RDA**	**R**	**6**
Rb	**11**	←	**Received data**	**104b**	**RDB**	**T**	**24**
	3						
	10						
Ta	**2**	→	**Transmitted data**	**103a**	**TDA**	**P**	**4**
Tb	**9**	→	**Transmitted data**	**103b**	**TDB**	**S**	**22**
	1		Frame ground		FG	A	1

7.5. Notice regarding open source code

The software of this product includes programs and libraries that are covered by the GNU General Public License (or "GPL"), available for example at following address: http://www.gnu.org/licenses/gpl.txt . Under this license, the source code for concerned elements is available on our Internet site ("Download" page); otherwise it can be obtained on request by e-mailing AETA AUDIO SYSTEMS (open_source@aeta-audio.com).

The software also includes the sipX stack covered by the LGPL license.

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NOTES

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